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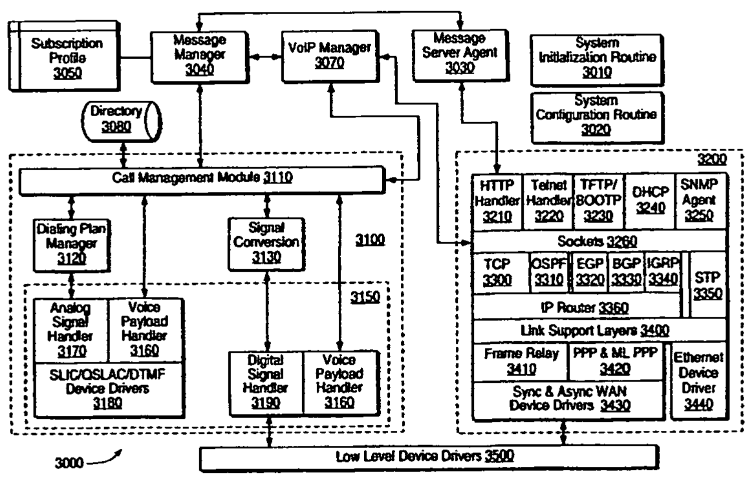

(57) Abstract

Disclosed are a method, apparatus, and system for transmission of informational messages to users of aural communication devices. An apparatus according to the disclosure provides a small key telephone system and an internetworking facility. The internetworking facility can retrieve messages from a message server for storage in the apparatus. The messages may then be transmitted to users of the aural communication devices under control of the small key telephone system. A message may be provided before or after a dial tone is provided when a user desires to make an outgoing call. Also a message may be provided before establishment of a communication path when a user receives an incoming call. Further disclosed is configuring central office switching equipment to perform as the disclosed apparatus. A still further disclosed aspect is configuring cellular communications systems to perform as the disclosed apparatus. In another disclosed aspect, computer telephony systems either based on POTS telephony, etherphone telephony, or voice over IP telephony, provide more varied messages that may include multimedia.

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TRANSMISSION OF AUDIO OR MULTIMEDIA INFORMATIONAL MESSAGES TO USERS

BACKGROUND OF THE INVENTION

FIELD OF THE INVENTION

The invention relates generally to telephony and more particularly to devices and methods that integrate telephony and other messaging systems.

BACKGROUND

Information exchange plays a vital role in modern society and one that is of growing importance. For decades the telephone system has played a central role in information exchange.

Increasingly, the functions of traditional telephony are converging with other means for information exchange, such as computer networks. One result of this is that users of equipment that integrates telephone and other functions have a richer set of communications possibilities available to them. Another result is that providers of networking services face increasing competition from businesses in many markets. A third result is that those who do not have the financial ability to purchase devices and services that provide integrated communications are at an even greater disadvantage in our society than in prior times.

Accordingly, it would be desirable for there to be an invention that provided a means for network service provides to enhance their revenues. Further, it would be desirable for there to exist a system that enabled access to telecommunication services to be subsidized. Still further, it would be desirable for a solution to these problems to

be able to take advantage of the increased functions of telecommunication devices, such as personal computers.

Conventional systems are lacking in features desirable for achieving these, and other, capabilities of the invention. For instance, conventional voice mail systems provide a means for messages to be provided to a telephone user; some even provide a means for users to retrieve messages based on user preferences. However, such systems are not suitable for network service providers to develop a revenue-generating service model that allows a sponsor's criteria to be consulted when determining whether and/or which informational message(s) should be sent to a subscriber. By providing a means for sponsors to customize the transmission of informational messages to aural communication device users, versions of the present invention provide features desirable to sponsors that are lacking in conventional systems and therefore provide increased revenue opportunities for network service providers.

Other conventional systems operate by a user dialing a "1-800" number to begin interacting with the system; the user then typically navigates a menu of options, enters an identification number, is transmitted messages, and thereafter may begin to make their desired call. Such systems present unnecessary inconvenience to the user. It will be appreciated that from a sponsor's perspective, increased user time, effort, and confusion in interacting with a system for transmitting informational messages may foster negative associations in the user's mind with the sponsor's goods, services, etc. It is therefore desirable that user frustration be minimized; accordingly features of the present invention can archive a sponsor's goals of transmitting informational messages to aural communication device users without a user having to actively remember and dial a "1-800" to enter the system. Such "1-800" number-type systems effectively demand that a user complete two telephone calls in order to participate in the system. Particularly for telecommunications systems that cater to busy customers, such as cellular systems, this additional time spent to make what is, in effect, an additional call, may be a barrier to wide adoption of the system. Still further, such "1-

800" number systems lack the desirable feature of the present invention by which an informational message may be transmitted to a user of a cellular communication device during the interval of time after the user has entered the number to which they wish to be connected but before the communications path is established.

The aforementioned capabilities and other objects are obtained by a method, apparatus, and system for the transmission of audio or multimedia informational messages, such as advertisements or the like, to aural communication device users.

SUMMARY

An illustrative aspect of the present invention involves practicing a method for providing at least one informational message to an aural communication device user in connection with an outgoing communication. One such method includes: detecting a first aural communication device is in use; retrieving a first informational message ; transmitting the first informational message to the first aural communication device for perception by the user; and providing an indication to the first aural communication device that the first aural communication device is available for connection to a second aural communication device. The aural communication device may be, for instance, a conventional phone, a computer-telephony system, a packetized voice communication device (e.g. a Voice over IP phone), a computer programmed to provide a Voice over IP client, or a wireless communication device, e.g. a cellular phone. The informational message may be transmitted before or after (or both) a dialing indication is provided to the aural communication device. Similarly, when a communication is incoming, informational messages may be transmitted to a user of the aural communication device after they place their aural communication device 'off hook' but before the incoming call is connected. The informational message may be in audio form, textual form, or multimedia form according to the situation and the capabilities of the aural communication device. Informational messages used with some versions of the invention may be stored in

compressed form and be decompressed before transmission; similarly, informational messages used with some versions of the invention may be stored in text form and be converted to an audio form before transmission.

Another illustrative aspect of the invention is that the informational messages sent to the aural communication devices can be controlled based on a profile of a user, a profile of a sponsor, or both. A user's profile may include, for instance positive or negative preferences, e.g., indications of the desire to be provided informational messages of a particular character or of the desire to not be provided informational messages of a particular character. A sponsor's profile may include, for instance, demographic, psychographic, geographic, or socio-economic preferences, e.g., indications of particular attributes of subscribers to be targeted for transmission of informational messages.

Still another illustrative aspect of the invention is identity verification of a user. Versions of the invention may prompt a user to identify himself or herself by, for instance, a personal identifier such as a PIN, or by a voice print. User identification information may be used in conjunction with user or sponsor profile(s) to provide a more customized selection of informational messages for the user. Reliable user identification may also operate with versions of the invention which invite commercial transactions and the user's identification used in conjunction with payment arrangements.

Another aspect of some versions of the present invention provides interactive informational messages ("feedback"). Feedback messages may present a user with selectable options. The user may select from the selectable options and have an output generated automatically. The output may be another message, possibly an electronic mail message, or another informational message. In some versions of the invention, the automated output may be a coupon that is sent to the user. The coupon may be sent to a merchant or other party for redemption by the user. In some versions

a selectable feedback option may be to be connected to another party such as a sponsor, merchant, etc. for additional information, electronic commercial transactions, etc. In other versions of the invention a selectable feedback option may be to establish a connection to a network resource, such as a web site, of a sponsor, merchant, etc.

A still further illustrative aspect of the present invention involves practicing an apparatus for providing at least one informational message to a user of an aural communication device. An illustrative apparatus is configured for connection to at least one aural communication device and a data network, said apparatus and includes: a processor; a memory comunicatively coupled with the processor; and a network interface communicatively coupled with the processor. The network interface is configured for communication over the data network. The illustrative apparatus further includes a subscriber line interface communicatively coupled with the processor and a message agent, the message agent adapted to retrieve at least one informational message for storage in said memory. The at least one message may be communicated to the subscriber line interface under the control of the processor. Such an apparatus provides the functions of a telephone system supporting several phone lines and an client operable with an internet-type network through which informational messages may be conveniently and flexibly retrieved thought client-server communications with a message server. Such an apparatus may, for instance, be placed at a customer's premises or collocated with a central office switch.

Another aspect of a version of the present invention involves configuring a central office switching system. An illustrative systems for providing informational messages to a user of aural communication device may include: a central switching system, the switching system routing voice telecommunications between at least a first aural communication device and a second aural communication device. The central switch is further configured for communicative coupling with a data network, and the system further includes a message server, the message server configured for

client-server communications with the switching system via said data network. The message server stores an informational message and the system further includes a message manager, the message manager configuring the switching system to request the informational message from the message server. The illustrative system further includes a call manager, the call manager configuring the switching system to perform steps comprising: receiving the informational message from said message manager; and transmitting the informational message to the first aural communication device.

Aspects of other versions of the invention may operate in a wireless communications network and provide a method for providing a textual, image, or multimedia informational message in conjunction with an incoming communication for an aural communication device. The aural communication device may have an associated identifier. Practicing an illustrative method according to the invention includes: receiving an indication of the incoming communication for the aural communication device; retrieving an informational message from a message server; establishing a data communication connection to the aural communication device; transmitting the informational message to the aural communication device; and thereafter establishing a connection to the aural communication device for the incoming communication. Retrieving an informational message from a message server may comprise: providing the identifier of the aural communication device to a message server; selecting an informational message responsive to the identifier; and retrieving the informational message selected responsive to the identifier. Selecting an informational message responsive to the identifier may also comprise: querying a subscription profile, the subscription profile comprising profile information for a subscriber, the subscriber associated with the identifier of the aural communication device; selecting an informational message responsive to the profile information; and retrieving the informational message selected responsive to the profile information.

Other illustrative aspects of the invention operable with cellular wireless communications provide a system for providing an informational message to a user of

a cellular aural communication device having an associated identifier. The illustrative system includes: a mobile switching system; a subscriber register, the subscriber register communicatively coupled with the mobile switching center, the subscriber register storing subscriber information, the subscriber register configured to receive a subscriber validation request from the mobile switching system and communicate a validation response to the mobile switching system, the validation response comprising a subscriber identifier; and a message server configured for client-server communication with the mobile switching system, the message server storing said informational message, wherein, in response to the initiation of an outgoing communication to a first phone number from the cellular aural communication device, the mobile switching system submits a validation request to the subscriber register, the mobile switching system receives a validation response from the subscriber register, the mobile switching system communicates a subscriber identifier to the message server, the message server communicates the informational message to the mobile switching system for transmission to the aural communication device.

Aspects of other versions of the invention may operate in a packet voice network in which the invention provides a method for providing informational messages to a user of aural communication device based on, for instance, voice over IP technology. Practicing an illustrative system includes: a voice over IP gateway resident in a central office or with head-end equipment in a cable network. The IP gateway routes voice communications between at least a first aural communication device, such as an IP phone, and a second aural communication device, such as a plain old telephone or an IP phone. The IP gateway is further configured for communicative coupling with a data network. Such a system further includes a message server, a message manager, and a call manager. The message server is configured for client-server communications with said gateway system via said data network, the message server further storing an informational message. The message manager configures the gateway system to request the informational message from the message server. The call manager configures the gateway system to perform steps

comprising: receiving said informational message from the message manager; and transmitting said informational message to the first aural communication device.

As will be further appreciated by this disclosure versions of the present invention provide a flexible integration of telephony and internetworking systems for the provision of informational messages to users of aural communication devices. Features of the invention allow rich messaging capabilities to be provided to users of computer telephony systems. Further, when informational messages used with the invention are advertisements or the like, versions of the invention provide a desirable means for networking service provides to increase their revenue by selling the opportunity to provide advertisements to aural communication device users. Such versions of the invention provide a service method and system which could greatly expand the access to aural communication technology for those of more modest means as service provides may subsidize the cost of providing telephony equipment and/or services with revenues gained from the provision of informational messages to aural communication device users.

BRIEF DESCRIPTION OF THE DRAWINGS

These and other features, aspects, and advantages of the present invention will become better understood with regard to the following description, appended claims, and accompanying drawings where:

Fig. 1A depicts a exemplary network architecture in which the invention may operate in accordance with an illustrative embodiment;

Fig. 1B depicts a exemplary public switched telephone network in which the invention may operate in accordance with an illustrative embodiment;

Fig. 1C	depicts additional features of the exemplary network architecture in which the invention may operate in accordance with an illustrative embodiment;
Fig. 1D	depicts an exemplary cellular network architecture in which the invention may operate in accordance with an illustrative embodiment;
Fig. 1E	depicts an exemplary network architecture including an interactive voice response system with which the invention may operate in accordance with an illustrative embodiment;
Fig. 2	depicts a block diagram of a hardware architecture in accordance with an illustrative embodiment;
Fig. 3	depicts a block diagram of a software architecture in accordance with an illustrative embodiment;
Fig. 4	depicts a flow diagram of a method for outgoing communication message transmission in accordance with an illustrative embodiment;
Fig. 5A	depicts a flow diagram of a method for communicating an AudioMT message in accordance with an illustrative embodiment;
Fig. 5B	depicts a flow diagram of a method for communicating a Feedback AudioMT message in accordance with an illustrative embodiment;
Fig. 6A	depicts a flow diagram of a method for communicating an AudioTM message in accordance with an illustrative embodiment;

Fig. 6B	depicts a flow diagram of a method for communicating a Feedback AudioTM message in accordance with an illustrative embodiment;
Fig. 7	depicts a flow diagram for a method for communicating an AudioICM message in accordance with an illustrative embodiment;
Fig. 8A	depicts a flow diagram for a method for communicating an AudioTM message to a VoIP client or Etherphone in accordance with an illustrative embodiment;
Fig. 8B	depicts a flow diagram for a method for communicating an AudioICM message to a VoIP client or Etherphone in accordance with an illustrative embodiment;
Fig. 9	depicts a flow diagram of a method for communicating an AudioTM message in accordance with an illustrative cellular embodiment;
Fig. 10	depicts a flow diagram of a method for communicating a TextICM message in accordance with an illustrative cellular embodiment;
Fig. 11	depicts a flow diagram of a method for communicating informational messages to a PC & phone in conjunction with an outgoing call in accordance with an illustrative embodiment;
Fig. 12	depict a flow diagram of a method for communicating informational messages to a PC & phone in conjunction with an incoming call in accordance with an illustrative embodiment;

Fig. 13A depicts a flow diagram of a method for communicating an PopupM message to a VoIP client in conjunction with an outgoing call in accordance with an illustrative embodiment;

Fig. 13B depicts a flow diagram of a method for communicating a PopupM message to a VoIP client in conjunction with an incoming call in accordance with an illustrative embodiment;

Fig. 14 depicts a flow diagram of a method for communicating a feedback PopupM message to a computer/phone in conjunction with an outgoing communication in accordance with an illustrative embodiment;

Fig. 15 depicts a flow diagram of a method for communicating a feedback PopupM message to a computer/phone in conjunction with an incoming communication in accordance with an illustrative embodiment;

Fig. 16 depicts a flow diagram of a method for alias dialing in accordance with an illustrative embodiment.

DETAILED DESCRIPTION OF THE INVENTION

NOTATIONS AND NOMENCLATURE

AudioMT An audio message that may be sent to an "ear piece" of an aural communication device before dial tone.

AudioTM An audio message that may be sent to an "ear piece" of an aural communication device after or during dial tone.

AudioICM	An audio message that may be sent to an "ear piece" of an aural communication device after an incoming call has been received.
TextICM	A Textual message that may be sent to a text display of an aural communication device before an incoming call; may also include image data.
Ring Back Tone	A special tone sent to a call originator that a destination aural communication device is ringing.
SLIC	Subscriber line interface circuit.
CPE	A customer premise equipment such as router, phone, key telephone system, FAD, modem, or broadband Integrated Access Device (IAD) etc. These devices may be used to connect subscribers to service providers' networks.
PopupM	A window or other display element in a computer system comprising a multimedia message.
Subscriber	A user who subscribes to a particular service provided by a service provider, e.g. a cellular phone subscriber, a Internet subscriber.
VoIP	"VoIP" stands for Voice over IP (Internet Protocol). It refers to a particular type of packetized voice communication technology that carries voice over packet oriented network(s) using the IP.
VoIP Gateway	A Gateway which allows the devices using VoIP technology to inter-operate with traditional telephones and vice versa.

Etherphone	"Etherphone" refers to a device that behaves like a telephone, but it is connected to a packet transport medium. Typically, the Internet Protocol is used; etherphones typically use a VoIP protocol for voice communication. Also known as IP phone.
Multimedia message	In addition to a standard meaning, the term 'multimedia message' includes, without limitation, single-media non-audio messages.
DSLAM	Digital Subscriber Line Access Multiplexor.

DESCRIPTION OF FIGURES

Fig. 1A depicts a exemplary network architecture **1000** in which illustrative embodiments invention may operate. An internet-type network **1100** is shown. Illustrative embodiments of the invention work with the Internet. Other networks using the Internet Protocol ("IP"), either public or private may also be used, as may networks which do not use the IP.

Addressable in the internet-type network **1100** are a first message server **1110** and a second message server **1120**. The first message server **1110** and the second message server **1120** generally function, at least, to store text, audio, or multimedia messages that may be transmitted in accordance with versions of invention. Further, the first message server **1110** and the second message server **1120** may provide management and control of the messages sent to subscribers. In some embodiments of the invention they gather statistics respecting messages transmitted to subscribers. The first message server **1110** or the second message server **1120** may be computing machinery of a general nature configured to act as a server in client-server communications. An illustrative server may be a SPARC-based workstation from SUN MICROSYSTEMS of Mountain View, CA running the SOLARIS operating

system and the Apache HTTPd server software, e.g., a "web server." Conveniently, the first message server **1110** or the second message server **1120** may be located at a service providers' Point of Presence. One of skill will recognize that versions of the invention may operate with one or several servers.

Further illustrated is a public switched telephone network **1200**, commonly known as "PSTN." Shown as part of the public switched telephone network **1200** are a first edge switch **1210** and a second edge switch **1240**. The first edge switch **1210** and the second edge switch **1240** typically are able to route voice calls to the public switched telephone network **1200** and internet traffic to the internet-type network **1100** including, for example, to the first message server **1110** or the second message server **1120**. As part of the public switched telephone network **1200**, the first edge switch **1210** is connected with a first central office switch **1220** and the second edge switch **1240** is connected with a second central office switch **1230**. Exemplary structures for the first edge switch **1210** or the second edge switch **1240** include, for example, Intelligent DSL Multiplexors, Multiservice Switches, ATM Edge Switches, Cable Modem Head-end Equipment and analogous equipment recognized as suitable in the art. Structures suitable for the first central office switch **1220** or the second central office switch **1230** include, for example, a carrier grade PBX switch and/or SS7 switch.

Referring now to the first edge switch **1210** as illustrative of the second edge switch **1240** as well, the first edge switch **1210** is connected to a first CPE device **1400**. The first CPE device **1400** may be an apparatus embodying aspects of the invention as will be described below. Typically functions of the first CPE device **1400** in accordance with illustrative embodiments include, access to the public switched telephone network **1200** and the internet-type network **1100** and control of one or more phone lines.

Shown connected with the first CPE device **1400** are a first set of aural communication devices **1500** including a first POTS phone **1510**, a first coupled PC & phone **1520**, and a first VoIP client **1530**. The first coupled PC & phone **1520** or the first VoIP client **1530** may communicate with the first CPE device **1400** via a first LAN **1540**. Any of the first set of aural communication devices **1500** may be used with the first CPE device **1400** in accordance versions of the invention as will be described below.

Also shown are a second set of aural communication devices **1600** including a second POTS phone **1610**, a second coupled PC & phone **1620**, and a second VoIP client **1630**. The second coupled PC & phone **1620** and the second VoIP client **1630** may communicate with a second CPE device **1300** via a second LAN **1640**. A first user may use one of the first set of aural communication devices **1500** to initiate a call to, or receive a call from, a second user operating one of the second set of aural communication devices **1600**. During the period in which the first user and the second user operate their respective aural communication devices, either or both may be transmitted an informational message by their respective CPE devices in accordance with versions of the invention as will be described below.

Fig. 1A also depicts a set of aural communication devices directly connected to a central office switch **1700** including a third POTS phone **1710** which also may be transmitted an informational message in some versions of the invention.

In some versions of the invention, the first CPE device **1400** is an integrated access device ("IAD") while in others it may be that, for example, the first central office switch **1220** has been suitably modified to perform functions and embody features of the invention. One of skill in the art having the benefit of this disclosure will recognize numerous structures of programmed or programmable logic capable of being suitably configured to perform functions according to versions of the invention.

Some versions of the invention could readily operate with VoIP Gateways, and it is contemplated that the invention will be independent of the underlying protocols used with the VoIP devices and VoIP Gateways **Fig. 1C** depicts additional features of the exemplary network architecture in which the invention may operate in accordance with illustrative embodiments. For clarity of organization, some features of the exemplary network architecture **1000** described in connection with **Fig. 1A** are also depicted in **Fig. 1C**. **Fig. 1C** further illustrates a first VoIP gateway **1800** and a second VoIP gateway **1825** that are communicatively connected with the first central office switch **1220** and the second central office switch **1230**, respectively. Connected between the VoIP Gateway and the subscribers may be a transport medium multiplexor, such as a DSLAM, Modem Banks, or Cable Head-end equipment. Alternatively, these functions may be integrated in the VoIP Gateway. For convenience, a DSLAM is used in this illustrative embodiment, as the particular transport medium multiplexor is not fundamental to the illustrative embodiment. A first DSLAM **1850** and a second DSLAM **1875** are also depicted. The first DSLAM **1850** may communicate with the first VoIP gateway **1800** and the first message server **1110**; the second DSLAM **1875** may communicate with the second VoIP gateway **1825** and the second message server **1120**.

Illustrative VoIP embodiments may be appreciated with reference to **Fig. 1C** by one of skill in the art who, also having the benefit of this disclosure, will apprehend how various other embodiments could be practiced. A converting computer **1555** is depicted as well as a POTS phone connected through a converting computer **1550**. The converting computer **1555** may be configured to convert the signaling of a POTS phone to VoIP transmissions for communication to the first VoIP gateway **1800** via the first DSLAM **1850**.

Also depicted is a first VoIP client programmed computer **1560**. The first VoIP client programmed computer **1560** may be connected to a first DSL equipment **1565** and the first DSL equipment **1565** may connect to the first DSLAM **1850**. The

first DSL equipment **1565** may be a DSL modem, or suitable customer premises equipment configured to perform the functions of a DSL modem. A user may use a speaker and microphone operatively coupled with the first VoIP client programmed computer **1560** to engage in VoIP communications according to many software and communications protocol configurations. Some embodiments use the NetMeeting application from Microsoft Corp. of Redmond, WA, U.S.A. that may employ the H.323 protocol, many other suitable applications are commercially available. Additionally depicted is a VoIP PC & phone **1570** connected to the first DSL equipment **1565**. Elements of the VoIP PC & phone **1570** may be used separately or in conjunction.

In some embodiments, a VoIP phone **1665** is connected to a second DSL equipment **1660**. The second DSL equipment **1660** is, in turn, connected to the second DSLAM **1875**. In other embodiments, a POTS phone **1670** is connected to a VoIP converter **1675** that converts the signaling to VoIP transmissions. The VoIP converter **1675** may then be connected to the second DSL equipment **1660**. A second VoIP programmed computer **1680** is also depicted and illustrates that various embodiments of the invention may be employed with one, or plural, parties to a aural communication communicating with VoIP systems.

Fig. 1D depicts an exemplary cellular network architecture **1900** in which the invention may operate in accordance with an illustrative embodiment. A mobile unit **1905** is shown. The mobile unit **1905** may be, for instance, a conventional cellular phone that supports text communication, personal digital assistant or computer with cellular communication capabilities, or other cellular-type communication device that supports text communication.

Several base transceiver stations **1903** provide a connection interface for the mobile unit **1905**. The base transceiver stations **1903** are controlled by a base station controller **1910** ("BSC") that is responsible for call hand over operations. Further

depicted is a mobile switching center **1920** ("MSC") that is responsible for setting up, managing, and clearing connections, as well as routing calls to an appropriate cell site. The mobile switching center **1920** also provides linkage to a signal transfer point **1980** ("STP") that may use SS7 protocols to provide signaling information required to establish circuit connections and disconnection with the public switched telephone network **1200**, as well as database retrieval from, and data sharing with, another entity. The mobile switching center **1920** communicates with cell sites through the base station controller **1910**.

The mobile switching center **1920** further communicates with a home local register **1930** ("HLR"), a visitor location register **1950** ("VLR"), and an operation and maintenance center **1940** comprising an equipment identity register **1960** ("EIR") and an authentication center **1970**. The home local register **1930** is a database that stores the subscriber information for all subscribers within the home service area of the service provider. The visitor location register **1950** stores information about visiting subscribers who are not in their home service area including, for example, roaming number information, so that subscribers may use their cellular phones while in another city. The equipment identity register **1960** stores the identification serial number of all cellular telephones activated within the coverage area while the authentication center **1970** stores a security key embedded into cellular phones. Both the equipment identity register **1960** and the authentication center **1970** are used for user authentication, activation and detection, and other operational and maintenance purposes.

Further depicted in **Fig. 1D** is a message server **1990** such as the first message server **1110** or the second message server **1120**. The message server **1990** may be collocated with the home local register **1930** so that when a user places a call from the mobile unit **1905** an identifier, such as the serial number in a cellular phone or phone identification, is transmitted from the mobile unit **1905** to one of the base transceiver stations **1903**. This information is then passed to the mobile switching center **1920**.

When the message server **1990** is collocated with the home local register **1930**, The mobile switching center **1920** may also communicate with the message server **1990** after the mobile switching center **1920** validates the calling phone number with the home local register **1930**. Operation of aspects of the invention that may employ the exemplary cellular network architecture **1900** is discussed in greater detail below in connection with **Fig. 9** and **Fig. 10**.

Various embodiments of the invention provide a telephone switching systems such as, for instance, key telephone switching systems, PBX systems, VoIP Gateways or the like. In accordance with various embodiments of the invention, a conforming telephone switching system may operate, for instance, with a PSTN, a Cellular network, or a Cable Network. One embodiment illustrative of various features of the invention is a device that has the characteristics of a key telephone system, an Internet access router, and a voice over IP gateway. However, to one of skill in the art, the teachings of this disclosure set forth other embodiments as well; and various alterations, modifications, omissions of elements are also within the grasp of one skilled in the art having the benefit of this disclosure. As one of skill in the art will recognize from the following descriptions, an apparatus with hardware and software architecture such as that of the illustrative versions of the invention described below may be an IAD. In addition to 8 telephone connections, and an Ethernet LAN connection towards the customer side the IAD may have a T1/E1 interface or a HDSL interface to a network. The apparatus may act as a small key telephone system, providing all the necessary functions of conventional plain-old telephone usage and a router for access to the internet-type network **1100**.

As noted, in some embodiments the first CPE device **1400** is an IAD and **Fig. 2** illustrates a block diagram of a hardware architecture of an illustrative IAD embodying aspects of the invention. The hardware architecture **2000** includes a microprocessor subsystem **2100**, a memory subsystem **2200**, a network interface subsystem **2300**, a set of subscriber interfaces **2400**, and a set of user interfaces **2500**.

The microprocessor subsystem **2100** comprises a MPC860 microprocessor **2110**, an address bus **2120**, a data bus **2130**, a memory controller **2140** and an DMA controller **2150**. The microprocessor subsystem **2100** typically also includes conventional supporting circuitry including, for example, a power, reset, debug port, and clock circuits (not shown). The MPC860 microprocessor **2110** is a commercially available MPC860 microprocessor from Motorola. Conveniently, the MPC860 includes an integrated Ethernet Media Access Controller ("MAC"). Embodiments that use the MPC860 conveniently configure the data bus **2130**, the memory controller **2140**, and the DMA controller **2150** according to Motorola's SAMBA Reference Design Board for the MPC860 processor.

The memory subsystem **2200** is communicatively coupled with the microprocessor subsystem **2100**. The memory subsystem **2200** comprises an DRAM subsystem **2210** and a flash memory subsystem **2220**. The flash memory subsystem **2220** may be implemented with four Advanced Micro Devices 29F800 (1M x 8) flash memory devices arranged as a 32-bit wide memory bank. The MPC860 microprocessor **2110** may access the memory subsystem **2200** through a glueless interface consisting of signals from the address bus **2120** and the data bus **2130**, and control signals from the memory controller **2140**. These signals are connected to the flash memory devices through a set of 3.3V-5V voltage translation buffers (not shown), to alleviate signal loading on the MPC860 microprocessor **2110**. The MPC860 microprocessor **2110** accesses to the flash memory subsystem **2220** are controlled by chip-select CS0 of the memory controller **2140**.

The network interface subsystem **2300** is communicatively coupled with the microprocessor subsystem **2100**. The network interface subsystem **2300** includes a field programmable gate array **2310** and a set networking ports **2350** that includes an HDSL port interface **2320**, a T1/E1 port interface **2330**.

The field programmable gate array **2310** manages multiplexing and routing of voice and data streams (or combined voice and data streams) over one or more of the set networking ports **2350**. The T1/E1 port interface **2330** may be implemented with a Dallas Semiconductor DS2152 Enhanced T1 Transceiver device available from Dallas Semiconductor of Dallas, Texas, U.S.A.. The DS2152 may be configured to support a variety of formats such as D4, ESF, and SLC-96. An external processor such as the MPC860 microprocessor **2110** accesses the DS2152's internal control and status registers through its parallel bus interface (not shown). The interface to the MPC860 microprocessor **2110**'s bus may be in the Motorola non-multiplexed mode. In alternative embodiments in which an E1 interface is desired, the T1/E1 port interface **2330** may be implemented with a Dallas Semiconductor DS2154.

A suitable structure for the HDSL port interface **2320** is an HDSL chip set from Metalink that comprises one MtH2430BL, two MtH2410AL and two MtH2445 processors that may be designed according to the Metalink reference design (all available from Metalink, Ltd. of Tel Aviv, Israel).

In some embodiments an ADSL port interface may be added to the set networking ports **2350**. Many suitable ADSL chip sets are commercially available, one example is from Motorola. One of skill will recognize many other commercially available suitable structures and designs.

In some embodiments a Cable Modem interface may be added to the set networking ports **2350** (or optionally in lieu of the HDSL port interface **2320**). A suitable Cable Modem interface may be implemented with, for instance, the BCM330-QAMLINK chipset from Broadcom Corp. of Irvine, California. Conveniently, in embodiments comprising a Cable Modem interface, a device so configured may communicate with Cable head-end equipment. When configured for communication with Cable head-end equipment features of the invention may conveniently be embodied in a 'set-top box'.

The set of user interfaces **2500** comprises a set of subscriber interfaces **2400**, a RS-232 port **2510**, an ethernet transceiver **2430**, and a UTP port **2440**. The set of subscriber interfaces **2400** comprises a pair of QSLACs **2410**, a set of four SLICs **2420**, and a set of four DTMF detectors **2450**. The set of user interfaces **2500** may provide the function of a "plain old telephone system" (POTS) interface, an Ethernet LAN interface, and an RS-232 interface.

The POTS interface function may be implemented by the pair of QSLACs **2410** which may be Advanced Micro Devices Am79Q02 QSLAC. Each of the pair of QSLACs **2410** may drive a set of four SLICs **2420**. The set of four SLICs **2420** may be Advanced Micro Devices Am7920 SLICs (available from Advanced Micro Devices, Sunnyvale, California, U.S.A). The set of four SLICs **2420** are connected to the set of four DTMF detectors **2450** and a power supply (not shown). Each of the set of four SLICs **2420** provides DC current to send voice signals to a telephone (not shown) connected with a two-wire line. Further each of the set of four SLICs **2420** manages the TIP and Ring leads of the telephone line interface, detects the off-hook status of a connected telephone, and works with the power supply and a ringing relay to switch the ringing signal on or off to the connected telephone. Each of the pair of QSLACs **2410** processes digital PCM voice data into analog signals and inputs them to one or more of the set of four SLICs **2420**. In the transmit path, the analog output of one of the set of four SLICs **2420** is processed by the QSLAC device and output in serial digital format to the PCM interface.

The Ethernet LAN interface may be implemented with a LXT970 10/100 ethernet transceiver **2430** available from Level One Communications of Sacramento, California, U.S.A. Conveniently the ethernet transceiver **2430** may be interfaced with the integrated MAC of the MPC860 microprocessor **2110**. The ethernet transceiver **2430** communicates through the UTP port **2440**. The design of the LXT970's analog interface to the UTP port **2440** may be as in the LXT970's databook. The LXT970 ethernet transceiver **2430** interfaces to the MPC860 microprocessor **2110** via its

Media Independent Interface, (MII). The MPC860 microprocessor **2110** MII signals are implemented on its Port D pins and four other pins that are defined as spare pins on the MPC860 microprocessor **2110**.

The RS-232 port **2510** may be centered on a SMC1 communication port of the MPC860 microprocessor **2110**. The SMC1 port may be used as a low-speed serial link for a resident ROM monitor. The design of this port comprises a Maxim 3232 RS-232 transceiver that converts SMC1's transmit and receive signals, to RS-232 levels.

Also depicted is a digital signal processor **2600** communicatively coupled with the microprocessor subsystem **2100**. The digital signal processor **2600** may operate with versions of the invention to provide digital audio signal generation for particular messages.

Fig. 3 depicts a block diagram of a software architecture **3000** that may operate with the hardware architecture **2000** in an illustrative version of the invention . The software architecture **3000** includes a system initialization routine **3010**, a system configuration routine **3020**, a Key/PBX module **3100**, an internet communication module **3200**, a message server agent **3030**, a message manager **3040**, and a VoIP manager **3070**.

The system initialization routine **3010** performs conventional system initialization functions. The system configuration routine **3020** allows a user or network operators to configure parameters used in conjunction with operation of the device such as the phone line numbers, an IP address for the device, an access speed to the network, etc.

In addition the system configuration routine **3020** allows configuration of a subscription profile **3050**. The subscription profile **3050** may specify, for example, a service type for users associated with the aural communication device to which a

device embodying aspects of the invention may communicate informational messages. The subscription profile **3050** may control, for example, a number of times which an informational message will be transmitted to an aural communication device. The subscription profile **3050** may also comprise a profile for a sponsor or for a user associated with an aural communication device to which the device transmits informational messages. The system configuration routine **3020** may configure the subscription profile **3050** in addition to conventional parameters used in conjunction with operation of the device.

The internet communication module **3200** includes a set of WAN device drivers **3430**. The set of WAN device drivers **3430** include both synchronous or asynchronous communications protocols. The set of WAN device drivers **3430** communicate with a set of low level device drivers **3500** that drive the chipsets of the T1/E1 port interface **2330**, the HDSL port interface **2320**, or Cable Modem interface. The set of WAN device drivers **3430** passes data to a frame relay services module **3410** and a PPP / ML PPP services module **3420**. The frame relay services module **3410**, the PPP / ML PPP services module **3420**, and an Ethernet device driver **3440** provide link layer functions. A set of link support layers **3400** supports the link layer function and provides data to network layer modules such as an IP router module **3360** and an STP module **3350**. The STP module **3350** implements the Spanning Tree Bridging Protocol that provides an IEEE 802-compliant MAC layer bridging engine containing both port state control and frame filtering/forwarding. The IP router module **3360** implements the functions of a conventional Internet Protocol router. Suitable implementations are widely available commercially including from RouterWare of Newport Beach, CA, U.S.A. A TCP module **3300** provides transport layer functions and several routing protocol modules, such as an OSPF module **3310**, an EGP module **3320**, an BGP module **3330**, and an IGRP module **3340** exchange routing information from the IP router module **3360** with gateways in other systems.

A socket layer **3260** provides an IP address and "port" for application layer modules to communicate with lower layers. Exemplary application layer modules include an SNMP agent **3250**, a DHCP module **3240**, a TFTP/BOOTP module **3230**, a Telnet handler **3220**, and an HTTP handler **3210** that operate conventionally.

The HTTP handler **3210** operates with a message server agent **3030**. The message server agent **3030** communicates with a message server such as the first message server **1110** or the second message server **1120**. In one embodiment of the invention, when an IAD is to be installed, an installer provides the IAD with the URL of a message server. Upon the IAD powering up and establishing a valid network connection, the IAD communicates an HTTP Open Connection Request to the message server. The message server transfers one or more informational messages, to the IAD where they are stored, for example in the memory subsystem **2200**. The message server agent **3030** may also periodically communicate an HTTP Open Connection Request to the message server and next communicate an HTTP Post Request to transfer logs and/or statistics gathered by the IAD relating to the informational messages transmitted by the IAD to various aural communication devices. In some embodiments, the message server agent **3030** may initiate a request to open a connection with the message server responsive to status of the subscription profile **3050**. The message server agent **3030** may then transfer one or more informational messages to the IAD.

Turning now to the Key/PBX module **3100** of the software architecture **3000**, the Key/PBX module **3100** includes a call management module **3110**, a Key/PBX driver module **3150**, a dialing plan manager **3120**, and a signal conversion module **3130**.

The Key/PBX driver module **3150** works with hardware described in connection with **Fig. 2**. A set of device drivers **3180** operate with the pair of QSLACs **2410**, the set of four SLICs **2420**, and the set of four DTMF detectors **2450**. When the

set of device drivers **3180** detect an "off hook" condition for a phone connected to one of the set of four SLICs **2420**, the set of device drivers **3180** coordinates operation of a ringing circuit, assembles DTMF tone digits, and generates various tones. The dialing plan manager **3120** communicates with an analog signal handler **3170** and a voice payload handler **3160**.

The analog signal handler **3170** communicates with the set of four SLICs **2420** and the pair of QSLACs **2410** to provide analog signaling handling including, for instance, Key Telephone System ("KTS") signaling, and Off Premise Extension ("OPX") signaling. The analog signal handling typically includes handling of "off hook" status of a connected analog aural communication device, such as the first POTS phone **1510**, ringing of the phone, dialing tone generation, and ring back tone generation. In addition, the analog signal handler **3170** operates with the set of four DTMF detectors **2450** to translate the DTMF tones to dialing digits.

The voice payload handler **3160** programs the pair of QSLACs **2410** and assembles PCM information stream into packetized form. In addition, the voice payload handler **3160** may provide jitter buffering of the voice payload based on an estimated round trip delay. In some versions, the pair of QSLACs **2410** operates with the digital signal processor **2600** to provide echo cancellation in the audio pathway. In other versions, an echo cancellation chipset such as TECO3264 from Lucent Technologies of Murray Hills, New Jersey may provide echo cancellation. As will be described further below, some versions of the messages are decompressed and/or undergo text-to-speech conversion. In embodiments when decompression or text-to-speech conversions is required, the voice payload handler **3160** coordinates the decompression or text-to-audio conversion with the digital signal processor **2600** and the microprocessor subsystem **2100**.

The dialing plan manager **3120** provides conventional dialing management functions, including, assembling the number of dialing digits, managing an inter-digit

time-out timer, managing the total duration of dialing time, and managing the maximum number of dialing digits allowed.

The call management module **3110** generally manages the state of the incoming and outgoing calls and provision of informational messages. For instance, during an incoming call in the analog case, a user picks up a handset that triggers the analog signal handler **3170** to indicate that the phone is off-hook. This information is passed to the call management module **3110** via the dialing plan manager **3120**. The call management module **3110** may then consult a message manager **3040** with the line identification of the user's line to determine if an informational message should be communicated for perception by the user (as will be described in detail below).

If a call is an extension call to another analog line under the control of the system, the call management module **3110** generates a signal with the line identification to the analog signal handler **3170** to indicate there is a call for that particular line. This triggers the analog signal handler **3170** to exert ringing to the line. When a user answer the phone, the analog signal handler **3170** detects this phone as "off hook," and informs the call management module **3110** that the call has been answered. The call management module **3110** then connects the two lines and puts the path to voice payload transfer mode.

The call management module **3110** additionally may consult a directory database **3080** to determine how the call should be switched based on portion (or all) of the dialing digits. In some embodiments, the directory database **3080** is an internal pre-configured table that is used to route the phone calls based on hierarchy digits. However, this is not fundamental, and in other embodiments routing information could resides on a server accessed through a directory protocol including, for instance, the Lightweight Directory Access Protocol ("LDAP"). (When LDAP is used, a suitable protocol stack should be integrated, as one of skill will appreciate).

An IAD according to some embodiments of the invention supports TDM based access to PSTN including, for instance, T1 through the T1/E1 port interface **2330**. In these embodiments, the call management module **3110** may check, for instance, the directory database **3080** to determine if the call is should be routed to the PSTN. If the call should be routed to the PSTN via the T1 the T1/E1 port interface **2330** using, for instance, CAS signaling, the call management module **3110** allocates an appropriate time slot (if available) for the call and converts the analog call signaling to CAS signaling. Thereafter, the call management module **3110** places the call with the dialing digits as the phone number to the PSTN.

An additional aspect of some versions of the invention is routing of calls over the internet-type network **1100** as a Voice over IP ("VoIP") call. When VoIP call routing is used, the call management module **3110** communicates with a VoIP manager **3070**. The call management module **3110** passes the dialing digits to the VoIP manager **3070**.

The VoIP manager **3070** then initiates the call in accordance with a suitable VoIP standard. In some embodiments of the invention the H.323 VoIP standard is used as promulgated by the International Telecommunication Union ("ITU") of Geneva, Switzerland. If H.323 is desired, an H.323 protocol stack available from Radvision of Tel Aviv, Israel, may be used. The particular VoIP protocol used is not fundamental to the present invention and it is contemplated that the invention will operate with VoIP protocols other than H.323 (now known or later created) according to the particular commercial and/or technical needs of the situation. During a VoIP call, the call management module **3110** may maintain the call status state throughout the call, and interact with the VoIP manager **3070** appropriately.

The VoIP manager **3070** additionally manages the conversion of internal call control signaling to signaling format in compliance with a voice over IP standard. In some embodiments a Gatekeeper (as proposed in the ITU-T H.323 specification) is

not implemented. The directory database **3080** may be used in place of the Gatekeeper as a simple database for the address conversion from dialing digits to IP address format; LDAP access may also be used.

In some embodiments providing VoIP, the G.711 audio codec is used. In addition, some embodiments employ the User Datagram Protocol ("UDP") network layer for the speech payload transfer over the internet-type network **1100**.

The message manager **3040** may look at the line identification, which is passed to it by the call management module **3110**, and check with the subscription profile **3050** to determine whether an AudioMT should be sent to that particular line in the case of a call initiated through a phone connected to the IAD. In the event an incoming call is received from the PSTN to a line connected to the IAD, the call management module **3110** will determine the line identification based on the incoming call information. (E.g. in the case of DID, a specific number is associated with a specific line. If it is rotary, then the call management module **3110** will assign the call to a particular line that is free.) The line identification is sent to the message manager **3040**, which in turn checks with the subscription profiles to determine whether an AudioICM should be sent to that line. If needed, the message manager **3040** will retrieve the message and pass it to the call management module **3110** which in turn passes it to the voice payload handler **3160**.

The call management module **3110** implements message transmission features in various embodiments of the invention. Aspects of illustrative embodiments will be described in connection with **Fig. 4** through **Fig. 16** below. Briefly here, the call management module **3110** operates with a message manager **3040** to coordinate the transmission of informational messages to users of aural communication devices under control by an IAD.

As will be described in detail below, an apparatus according to illustrative versions of the invention may be, for example, the first CPE device **1400** and may

access message servers such as the first message server **1110** to retrieve informational messages which may be stored, for example in the DRAM subsystem **2210**, for later communication to, for example, one of the first set of aural communication devices **1500** under its control. Call management as described below allows the apparatus to transmit informational messages, including advertisements or the like, to aural communication devices under its control.

ILLUSTRATIVE EMBODIMENTS

Features of various versions of the invention will now be provided for further illustration. One of skill in the art will appreciate that the following embodiments are illustrative of the invention and that various modifications, omissions, and alterations may be made without departing from the scope and spirit of the invention as set forth in the appended claims. Reference will be made to **Fig. 1A–Fig. 1D** in describing illustrative embodiments with the understanding that the illustrative embodiments are not limited to the configurations of **Fig. 1A–Fig. 1D**.

1. PHONE CONNECTED TO PUBLIC SWITCH INFRASTRUCTURE THROUGH A CPE

In some embodiments of the invention the first POTS phone **1510** is connected to the public switched telephone network **1200** with the first CPE device **1400**. The first CPE device **1400** may be an IAD with hardware and software architectures as described in connection with **Fig. 2** and **Fig. 3**.

1.A. OUTGOING COMMUNICATION HANDLING

For purposes of illustration of outgoing communication handling, a first user operating the first POTS phone **1510** is assumed to be making an outgoing

communication to a second user operating one of the second set of aural communication devices **1600**.

Fig. 4 depicts a flow diagram of an 'outgoing communication message transmission' method **4000** in accordance with an illustrative embodiment. The first CPE device **1400** may implement aspects of the 'outgoing communication message transmission' method **4000**. Processing initiates at a 'start' terminal **4025** and one of the set of four SLICs **2420** generates an 'indication the first POTS phone **1510** is 'off hook' **4050**. The set of device drivers **3180** receive a signal form one of the set of four SLICs **2420** that is passed to an "off hook' detection' process **4075** in the analog signal handler **3170** that detects the signal and provides an indication thereof to the call management module **3110**.

Next, an 'AudioMT to first user' decision process **4125** involves the call management module **3110** consulting the message manager **3040** to determine whether an AudioMT should be sent to the first POTS phone **1510** at this time. The message manager **3040** examines the line identification, which is passed to it by the call management module **3110**, and may consult a subscription profile **3050** associated with the first POTS phone **1510** or the AudioMT to determine whether to send the AudioMT.

The subscription profile **3050** may comprise records commonly used by advertisers or marketers in customizing their messages including, for example, a frequency with which the AudioMT is to be communicated, a class or group of users to which the AudioMT is to be communicated, and temporal limits on when the AudioMT should be communicated. The temporal limits may indicate particular times during the day when the AudioMT should or should not be communicated, and may also indicate the expiry of a 'lifespan' associated with the AudioMT.

If the AudioMT should be communicated to the first aural communication device, the 'AudioMT to first user' decision process **4125** exits through its 'yes'

branch to enter an 'AudioMT retrieval' process **4150**. The 'AudioMT retrieval' process **4150** is performed by the message manager **3040** and retrieves the AudioMT from the DRAM subsystem **2210**.

Process flow continues to a 'feedback AudioMT' decision process **4175** performed by the message manager **3040**. Some embodiments of the invention operate with informational messages that offer feedback or interactive ("feedback") options to a user. In some embodiments, the informational messages convey to the user feedback options. For example, a user may hear, "Press star-1 to learn more; press start 2 to receive a coupon for this item, or press star-3 to speak with an operator to participate in a promotion . . . " Some embodiments include an indication in a header portion of the informational message indicating whether the informational message will present feedback options to the user. The 'feedback AudioMT' decision process **4175** detects whether the AudioMT retrieved in the 'AudioMT retrieval' process **4150** indicates that it will present feedback options. If so, the 'feedback AudioMT' decision process **4175** exits through its 'yes' branch and process flow continues to a 'feedback AudioMT communication' process **4300** that will be described below in connection with **Fig. 5B**. If the AudioMT retrieved in the 'AudioMT retrieval' process **4150** does not indicate that it will present feedback options, the 'feedback AudioMT' decision process **4175** exits through its 'no' branch and process flow continues to an 'AudioMT communication' process **4200** that will be described below in connection with **Fig. 5A**.

From either the 'AudioMT communication' process **4200** or the 'feedback AudioMT communication' process **4300** process flow continues to a 'remaining messages' decision process **4400** that consults the message manager **3040** and determines whether one or more additional AudioMTs should be sent. If additional AudioMTs should be sent, the 'remaining messages' decision process **4400** exits through its 'yes' branch and process flow returns to the 'AudioMT retrieval' process **4150** for another iteration. When no additional AudioMTs should be sent, the

'remaining messages' decision process **4400** exits through its 'no' branch and process flow continues to the 'outgoing call handling' process **4425**. The call management module **3110** performs the 'outgoing call handling' process **4425** and instructs the analog signal handler **3170** to provide an indication that the first POTS phone **1510** is available for connection to one of the second set of aural communication devices **1600**, typically a conventional dial tone. Ingress digits are typically then received from the first POTS phone **1510** and handled by the call management module **3110** to initiate a call.

Process flow continues to an 'AudioTM to first user' decision process **4450** that involves the call management module **3110** consulting to the message manager **3040** determine whether to send an AudioTM to the first POTS phone **1510** in a fashion analogous to that described above in connection with the 'AudioMT to first user' decision process **4125**. If the AudioTM should be communicated to the first aural communication device, the 'AudioTM to first user' decision process **4450** exits through its 'yes' branch to enter an 'AudioTM retrieval' process **4475**. The message manager **3040** performs the 'AudioTM retrieval' process **4475** that retrieves the AudioTM from the DRAM subsystem **2210**. Process flow continues to a 'feedback AudioTM' decision process **4500** that (similar to the 'feedback AudioMT' decision process **4175**) will detect if the AudioTM is a feedback AudioTM. If so, the 'feedback AudioTM' decision process **4500** exits though its 'yes' branch to enter a 'feedback AudioTM communication' process **4600**; if not, the 'feedback AudioTM' decision process **4500** exits through its 'no' branch and process flow enters an 'AudioTM communication' process **4700**. The 'AudioTM communication' process **4700** and the 'feedback AudioTM communication' process **4600** will be described in detail below in connection with **Fig. 6A** and **Fig. 6B**, respectively.

Process flow continues to a 'remaining messages' decision process **4800** that consults the message manager **3040** and determines whether one or more additional AudioTMs should be sent. If additional AudioTMs should be sent, the 'remaining

messages' decision process **4800** exits through its 'yes' branch and process flow returns to the 'AudioTM retrieval' process **4475** for another iteration. When no additional AudioMTs should be sent, the 'remaining messages' decision process **4800** exits through its 'no' branch and process flow continues to the 'call establishment handling' process **4825**.

The 'call establishment handling' process **4825** performs conventional steps to establish the communication path to one of the second set of aural communication devices **1600** familiar to one of skill in the art. Briefly, the dialing plan manager **3120** assembles the number of dialing digits and passes them the call management module **3110**. The call management module **3110** then checks the directory database **3080** to see how to route the call. Assuming, for example, the communication is to route through the T1/E1 port interface **2330** to the public switched telephone network **1200**, the call management module **3110** sets up a logical path and sends out a connection request through the call management module **3110**. The call management module **3110** determines an appropriate digital interface signaling protocol. If CAS signaling, the call management module **3110** assigns a time slot (DS0) and invokes signal conversion to initiate the ABCD bit according to the connection signaling protocol. If PRI ISDN interface, the call management module **3110** performs according to the PRI ISDN specification. If the remote is not busy, the call management module **3110** maintains the logical path, and establishes the voice payload transfer state. During the communication, the call management module **3110** passes back and forth the voice payload between the calling and called party. This typically includes the handshakes (such as hearing the remote phone is ringing, the remote handset is picked up – off hook) before the speech path is fully connected. If the remote is busy, the call management module **3110** generates a busy tone to the first POTS phone **1510**. If the number is invalid the call management module **3110** generates a fast busy tone. Process flow completes through an 'end' terminal **4900**.

The 'AudioMT communication' process **4200** will now be described in greater detail. **Fig. 5A** depicts a flow diagram of a method communicating an AudioMT in accordance with an illustrative embodiment. The 'AudioMT communication' process **4200** initiates at 'start' terminal **4205** and process flow continues to a 'compressed message' decision process **4220** performed by the voice payload handler **3160**. The 'compressed message' decision process **4220** exits through its 'yes' branch if the AudioMT is in compressed form and a 'decompression' process **4230** decompresses the AudioMT. Process flow continues to a 'text to audio' decision process **4240** that exits through its 'yes' branch if the AudioMT is in textual form and should be converted to audio. A 'text to audio conversion' process **4250** performs the conversion. The 'decompression' process **4230** and the 'text to audio conversion' process **4250** may operate conventionally with the digital signal processor **2600** to perform their functions. Process flow continues to an 'AudioMT transmission' process **4260** that transmits the AudioMT in audio form to the first POTS phone **1510** for perception by the first user. Process flow continues to a ''messages sent' flag setting' process **4280** that sets an internal flag indicating that a message has been sent and process flow completes through an 'end' terminal **4290**.

The 'feedback AudioMT communication' process **4300** will now be described in more detail with reference to **Fig. 5B**. Some embodiments of the invention support interactive informational messages (termed "feedback"). Process flow initiates at a 'start' terminal **4305** and continues to a 'compressed message' decision process **4315** performed by the voice payload handler **3160**. The 'compressed message' decision process **4315** exits through its 'yes' branch if the AudioMT is in compressed form and a 'decompression' process **4320** decompresses the AudioMT. Process flow continues to a 'text to audio' decision process **4325** that exits through its 'yes' branch if the AudioMT is in textual form and should be converted to audio. A 'text to audio conversion' process **4330** performs the conversion. The 'decompression' process **4320** and the 'text to audio conversion' process **4330** may operate conventionally with the digital signal processor **2600** to perform their functions. Process flow continues to a

'feedback AudioMT transmission' process **4335** that begins transmitting the AudioMT in audio form to the first POTS phone **1510** for perception by the first user.

Next, a 'AudioMT feedback indication' decision process **4340** detects if the message sent to the user requires a feedback from the user. If so, then the user is prompted to provide a feedback indication. For instance, an AudioMT may offer one or more feedback options along with associated indications the user may make to select among the one or more feedback options. Generally, the type, quality, and nature of feedback options are only limited by the capabilities of the aural communication device being used. Illustrative versions of the invention are described below; one of skill in the art will appreciate various modifications, alterations and adaptations of the illustrative embodiments that still lie within the spirit and scope of the invention as set forth in the appended claims.

In some illustrative versions, prompting for feedback input is similar to that found with conventional voice mail systems, e.g. "press 1 for to speak with an operator to learn more; press 2 to participate in a promotion; press 3 to receive a catalog; press 4 to proceed to make your original call . . . " Advertisers or other providers of informational messages may customize the particular message and feedback option prompting to their liking.

When a user provides the dialing sequence for a feedback option, one of the set of four DTMF detectors **2450** detects the tone sequence, converts it to binary and provides the binary sequence to the call management module **3110** . The 'AudioMT feedback indication' decision process **4340** exits through its 'yes' branch and the call management module **3110** couples the information with a line identity, and other information such as a message identifier, IAD identifier to form a feedback indication message. Process flow continues to a 'feedback indication storage' process **4350** that passes the feedback indication message to the message server agent **3030** for communication to the message server. Such information may be logged by message

servers, used to generate usage profiles of subscribers, used to determine pricing of informational messages, and other information collation, and customization operations with which one of skill in the art is familiar.

Process flow continues to a 'call redirection' decision process **4355**. Some AudioMTs may prompt the user to have a call placed ("redirected call") to a third party including, for instance, a sponsoring entity. Some embodiments place the redirected call upon receipt of the user feedback indication while others place the redirected call after the user completes their outgoing call.

The call management module **3110** upon receiving the user feedback selection determines if the selection is for a redirected call. In some embodiments, the AudioMT also includes one or more fields comprising selections and associated redirected call numbers. In other embodiments, the call management module **3110** may receive a response message from the message server comprising the number to dial for the redirected call. Still other embodiments use "alias dialing". With alias dialing the user may prompted to use a word, mnemonic, acronym, etc. created by the letters found on telephone keypad, e.g. "press star ITX to speak with the ITX Networks operator" where "star ITX" corresponds to the "*", "4", "8", "9" keys on a conventional telephone keypad. The alias may correspond to the number that should be dialed to complete the redirected call.

In still other embodiments, alias dialing may be used whenever the user begins using their aural communication device. In these embodiments, when the user places their aural communication device "off hook" and begins to enter a dialing sequence, the call management module **3110** in an IAD, CPE, or central office switch can detect an alias from the dialing sequence and place a call to a target number for which the alias is word, mnemonic, acronym etc. The conversion from the alias name to the target telephone number may be a simple table look up, or a directory call conversion or a message exchange with the message server. This feature allows a feedback

message to prompt a user, for instance, "Dial star ITX at any time to reach ITX Networks" and the user may pick up their aural communication device at some time and use this alias to reach ITX Networks. Further description of alias dialing is provided below with reference to **Fig. 16**.

Returning to **Fig. 5B**, the 'call redirection' decision process **4355** exits through its 'yes' branch if the call management module **3110** determines that the selected feedback involves a call being placed to a third party. A 'redirected call' process **4360** executes in which the call management module **3110** places the call.

If the 'feedback AudioMT transmission' process **4335** determines that the selected feedback does not involve a call being placed to a third party and process flow continues to an 'automated output' decision process **4365**.

The 'automated output' decision process **4365** determines whether the selected feedback option involves automated output generation. If so process flow continues to an 'output generation' process **4370**. The 'output generation' process **4370** involves the call management module **3110** posting to the message server the feedback selection and also comprises the message server generating the automated output.

In some embodiments, the automated output may be an email message sent to an address associated with the user of the aural communication device. The message may be, for instance, a promotion, coupon, or other information inviting a commercial transaction. The automated output may be mail sent by conventional postal methods, in which case the message server would provide an indication to, for instance, a human operator to send the mail. In still other embodiments, the automated output may be a coupon redeemable at a merchant. The message server may provide the information necessary to prepare the coupon directly to the merchant for the user to pick up from the merchant and redeem. The user may be provided (via their aural communication device, electronic mail, or otherwise) with an identification number or personal code to identify himself or herself at the merchant in order to redeem the

coupon . This feature of some embodiments of the invention allows the great volumes of conventional commercial mail in paper form sent to residences to be substituted with coupon provision according to embodiments disclosed herein. In still other embodiments, the automated output is not sent to the user but may be sent to another person. In yet other embodiments, the automated output may be a next informational message sent to the user's aural communication device. The next informational message may prompt the user to select from additional feedback options.

From the 'output generation' process **4370** process flow continues to an 'output communication' process **4375** in which the output from the 'output generation' process **4370** is communicated to the appropriate recipient. From the 'output communication' process **4375** or the 'redirected call' process **4360** process flow continues to a 'remaining feedback' decision process **4380**. The feedback options presented to a user may provide for a next feedback option selection after the user has selected a first feedback option and completed a first feedback process. If the 'remaining feedback' decision process **4380** detects that a next feedback option has been selected, it exits through its 'yes' branch and a 'next feedback message selection' process **4385** executes. The 'next feedback message selection' process **4385** involves the call management module **3110** consulting a message server for the next feedback message. Process flow continues to a 'next feedback message communication' process **4390** in which the next feedback message is transmitted to the aural communication device and process flow returns to the 'feedback indication storage' process **4350** for another iteration.

When the 'remaining feedback' decision process **4380** exits through its 'no' branch, indicating no remaining feedback options are to be presented, or if no feedback indication is required during the 'AudioMT feedback indication' decision process **4340**, process flow continues to a 'messages sent flag setting' process **4395** that sets an internal flag that the AudioMT has been sent and process flow completes through an 'end' terminal **4399**.

The 'AudioTM communication' process **4700** will now be described in greater detail. **Fig. 6A** depicts a flow diagram of a method communicating an AudioTM in accordance with an illustrative embodiment. Process flow initiates at a 'start' terminal **4705** and continues to a 'compressed message' decision process **4715**. If the AudioTM is compressed, a 'compressed message' decision process **4715** exits through its 'yes' branch and a 'decompression' process **4720** decompresses the AudioTM. Process flow continues to a 'text to audio' decision process **4725** that exits through its 'yes' branch if the AudioTM is in text form and should be converted to audio. A 'text to audio conversion' process **4730** performs the conversion. The 'text to audio conversion' process **4730** and the 'decompression' process **4720** may operate conventionally with the digital signal processor **2600** to perform their functions. Process flow continues to a 'AudioTM transmission' process **4740** that begins transmission of an AudioTM in audio form to the first POTS phone **1510** for perception by the first user.

Next, a 'destination ring back tone detected' decision process **4745** exits through its 'yes' branch if the ring back tone has been detected from one of the second set of aural communication devices **1600**. If so, a 'set Ring Back Tone flag' process **4750** sets a flag indicating that the ring back tone has been detected. Process flow continues to a 'destination answered' decision process **4755** that exits through its 'yes' branch if the call has been answered by one of the second set of aural communication devices **1600**.

If the 'destination answered' decision process **4755** exits through its 'yes' branch, a 'set Call Answered flag' process **4760** sets a flag indicating that the call has been answered. Process flow continues from here to a 'set Messages Sent flag' process **4770** that sets a flag indicating that a message has been sent and process flow completes through an 'end' terminal **4790**.

If the 'destination answered' decision process **4755** exits through its 'no' branch, process flow continues to a 'Ring Back Tone flag set' decision process **4765** that determines if the flag indicating that a ring back tone has been detected is set. If so, the 'Ring Back Tone flag set' decision process **4765** exits through its 'yes' branch and a mixed AudioTM and Ring Back Tone transmission' process **4780** acoustically mixes the ring back tone and the AudioTM for transmission to the first POTS phone **1510** and process flow returns to the 'AudioTM transmission' process **4740**. If the 'Ring Back Tone flag set' decision process **4765** determines that the flag indicating that a ring back tone has been detected is not set, it exits through its 'no' branch and process flow continues to a 'time expired' decision process **4775**. The 'time expired' decision process **4775** determines whether a time limit in which the line is permitted to ring has expired. No particular time limit is fundamental to the features of the invention. If the time limit has not expired, process flow continues to the 'AudioTM transmission' process **4740** to continue transmission of the AudioTM. When the time limit expires, the 'time expired' decision process **4775** exits through its 'yes' branch and process flow completes through the 'end' terminal **4790**.

The 'feedback AudioTM communication' process **4600** will now be described in more detail with reference to **Fig. 6B**. The 'feedback AudioTM communication' process **4600** is similar is the 'feedback AudioMT communication' process **4300**. Process flow initiates at a 'start' terminal **4605** and continues to a 'compressed message' decision process **4615** performed by the voice payload handler **3160**. The 'compressed message' decision process **4615** exits through its 'yes' branch if the AudioTM is in compressed form and a 'decompression' process **4620** decompresses the AudioTM. Process flow continues to a 'text to audio' decision process **4625** that exits through its 'yes' branch if the AudioTM is in textual form and should be converted to audio. The 'text to audio conversion' process **4630** performs the conversion. The 'decompression' process **4620** and the 'text to audio conversion' process **4630** may operate with the digital signal processor **2600**. Process flow

continues to a 'feedback AudioTM transmission' process **4635** that transmits the AudioTM in audio form to the first POTS phone **1510** for perception by the first user.

Unlike the 'feedback AudioMT communication' process **4300**, in the 'feedback AudioTM communication' process **4600**, when the 'feedback AudioTM transmission' process **4635** executes and user feedback processing occurs, the dial tone to the aural communication device conveniently is suppressed. This is indicated by a 'dial tone suspension' block **4637**.

Process flow within the 'dial tone suspension' block **4637** is similar to that in the 'feedback AudioMT communication' process **4300**.

A 'feedback indication' decision process **4640** detects whether the message sent to the user requires a feedback from the user, then the user will be prompted for the feedback. When a user provides the dialing sequence for a feedback option, one of the set of four DTMF detectors **2450** detects the tone sequence, converts it to binary and provides the binary sequence to the call management module **3110** . The 'feedback indication' decision process **4640** exits through its 'yes' branch and the call management module **3110** couples the information with a line identity, and other information such as a message identifier, IAD identifier to form a feedback indication message. Process flow continues to a 'feedback indication storage' process **4650** that passes the feedback indication message to the message manager **3040** for communication to the message server. Such information may be logged by message servers, used to generate usage profiles of subscribers, used to determine pricing of informational messages, and other information collation, and customization operations with which one of skill in the art is familiar. Process flow continues to a 'call redirection' decision process **4655** that exits through its 'yes' branch if the call management module **3110** determines that the selected feedback involves a call being placed to a third party. A 'redirected call' process **4660** executes in which the call management module **3110** places the call.

If the 'call redirection' decision process **4655** determines that the selected feedback does not involve a call being placed to a third party process flow continues to an 'automated output' decision process **4665**. The 'automated output' decision process **4665** determines whether the selected feedback option involves automated output generation. If so process flow continues to an 'output generation' process **4670**. The 'output generation' process **4670** involves the call management module **3110** posting to the message server the feedback selection and also comprises the message server generating the automated output. The automated output may be, for instance, of the type described above in connection with the 'feedback AudioMT communication' process **4300**.

From the 'output generation' process **4670** process flow continues to an 'output communication' process **4675** in which the automated output generated by the 'output generation' process **4670** is communicated to the appropriate recipient. From the 'output communication' process **4675** or the 'redirected call' process **4660** process flow continues to a 'remaining feedback' decision process **4680**. The feedback options presented to a user may provide for next feedback option selection after having selected a first feedback option and completed a first feedback process. If the 'remaining feedback' decision process **4680** detects that a next feedback option has been selected, it exits through its 'yes' branch and a 'next feedback message selection' process **4685** executes. The 'next feedback message selection' process **4685** involves the call management module **3110** consulting a message server for the next feedback message. Process flow continues to a 'next feedback message communication' process **4690** in which the next feedback message is transmitted to the aural communication device and process flow returns to the 'feedback indication storage' process **4650** for another iteration.

When the 'remaining feedback' decision process **4680** exits through its 'no' branch, indicating no remaining feedback options are to be presented, or if no feedback indication is required during the 'feedback indication' decision process **4640**,

process flow continues to a 'messages sent flag setting' process **4695** that sets an internal flag that the AudioMT has been sent. Process flow continues to a 'dial tone generation' process **4697** where the call management module **3110** issues a command to provide dial tone to the user. The user may then enter a dialing sequence and process flow completes through an 'end' terminal **4699**.

The various feedback and options and embodiments discussed above in connection with the 'feedback AudioMT communication' process **4300** are equally operable with the 'feedback AudioTM communication' process **4600**. That description is applicable to the 'feedback AudioMT communication' process **4300** and is provided by referring the reader to the discussion above.

1.B. INCOMING COMMUNICATION HANDLING

For purposes of illustration of incoming communication handling a second user operating the second POTS phone **1610** is assumed to be receiving an incoming communication from a first user operating one of the first set of aural communication devices **1500**.

Fig. 7 depicts a flow diagram of an AudioICM communication method **7000**. The second CPE device **1300** implements aspects of AudioICM communication method **7000** in this illustrative embodiment.

Process flow initiates at a 'start' terminal **7010** and continues to receive an 'incoming call' data block **7020** indicating that an incoming communication has been received from the public switched telephone network **1200**. Next, a 'ring and ring back tone generation' process **7030** activates one of the set of four SLICs **2420** in the second CPE device **1300** to ring the second POTS phone **1610**. The 'ring and ring back tone generation' process **7030** may also generate a ring back tone to the one of

the first set of aural communication devices **1500** initiating the incoming communication to indicate that the receiving phone is ringing.

Next, a 'call answered' decision process **7040** exits through its 'no' branch if the call has not been answered by the second POTS phone **1610** and a 'ring limit' decision process **7050** determines if a number of allowable rings has been exceeded. If not, the 'ring limit' decision process **7050** exits through its 'no' branch and process flow returns to the 'call answered' decision process **7040**. If the allowable number of ring has been exceeded, the 'ring limit' decision process **7050** exits through its 'yes' branch to enter a 'set No Answer flag' process **7060**. The 'set No Answer flag' process **7060** sets a flag indicating the call was not answered and the AudioICM communication method **7000** completes through an 'end' terminal **7070**.

If the call is answered by the second POTS phone **1610**, the 'call answered' decision process **7040** exits through its 'yes' branch and processing continues to an 'AudioICM to second user' decision process **7080**. The 'AudioICM to second user' decision process **7080** may consult a subscription profile to determine whether an AudioICM should be sent as has been described above in connection with AudioMTs and AudioTMs. If the AudioICM should not be transmitted, the 'AudioICM to second user' decision process **7080** exits through its 'no' branch and the AudioICM communication method **7000** completes through the 'end' terminal **7070**.

If an AudioICM is to be transmitted, the 'AudioICM to second user' decision process **7080** exits through its 'yes' branch and a 'retrieve AudioICM' process **7090** retrieves the AudioICM from the memory subsystem **2200**. Next, if the AudioICM is compressed, a 'compressed message' decision process **7100** exits through its 'yes' branch and a 'decompression' process **7110** decompresses the AudioICM. Process flow continues to a 'text to audio' decision process **7120** that exits through its 'yes' branch if the AudioICM is in text form and should be converted to audio. A 'text to audio conversion' process **7130** performs the conversion. The 'decompression'

process **7110** and the 'text to audio conversion' process **7130** may operate conventionally with the digital signal processor **2600** to perform their functions. Process flow continues to an 'AudioICM transmission' process **7140** that transmits the AudioICM in audio form to the second POTS phone **1610** for perception by the second user and begin a next iteration.

Process flow continues to a 'remaining messages' decision process **7150**. If a next AudioICM is not to be transmitted the 'remaining messages' decision process **7150** exits through its 'no' branch and the AudioICM communication method **7000** completes through the 'end' terminal **7070**. If a next message is to be transmitted the 'remaining messages' decision process **7150** exits through its 'yes' branch and a 'continue Ring Back Tone' process **7160** executes. In some versions, the 'continue Ring Back Tone' process **7160** provides the Ring Back Tone to the originator of the incoming communication. In some other versions, the 'continue Ring Back Tone' process **7160** sends a call connected message to the switching system to indicate that the incoming communication has been accepted and thereafter a message may be sent to the first user indicating the first user that his or her call is connecting with the second user. The message indicating that his or her call is connecting with the second user may be substituted by, or accompanied with, an informational message such as has been described above in connection with Part I.A. From the 'continue Ring Back Tone' process **7160** process flow returns to the 'retrieve AudioICM' process **7090** to retrieve the next AudioICM.

2. *VOICE OVER IP CLIENT, ETHERPHONE, OR REGULAR PHONE CONNECTED THROUGH CONVERTER TO FUNCTION AS A VOICE OVER IP PHONE CONNECTED THROUGH A VOIP GATEWAY*

2.A. *GATEWAY AT CUSTOMER PREMISES*

In some embodiments of the invention the first VoIP client **1530** is connected to the internet-type network **1100** through a gateway located at a customer's premises. In these embodiments a conventional gateway is modified to include programmed or programmable logic for the transfer of message files from a message server to the VoIP gateway (and suitable memory for storage of the transferred files). In addition, the call management module **3110** is modified to provide suitable incoming and outgoing communication handling of AudioTM and AudioICM messages. Handling of AudioMT messages is analogous to the processes described in connection with **Fig. 4, Fig. 5A**, and **Fig. 5B**.

For purposes of illustration of outgoing communication handling a first user operating the first VoIP client **1530** is assumed to be sending an outgoing communication to a second user operating one of the second set of aural communication devices **1600**.

Some embodiments of the invention handle VoIP communication transport (both incoming and outgoing). Some of these embodiments may comprise the hardware architecture **2000** of **Fig. 2** and the software architecture **3000** of **Fig. 3** in an IAD. Others of these embodiments may comprise software configuring conventional central office switching hardware or PBX systems to perform functions of (all or part of) the hardware architecture **2000** and the software architecture **3000**.

Some VoIP handling embodiments use the ITU H.323 protocol, although as noted above, the particular protocol is not fundamental. One of skill art, having the benefit of this disclosure, will appreciate how features and advantages of the invention may be obtained with other VoIP protocols as well. For illustration, H.323 Gatekeepers will be not be discussed as they, too, are not fundamental to practicing the invention.

Some VoIP handling embodiments perform functions of conventional VoIP Gateway communication handling. In some embodiments of the invention, ITU-T Q.931 is the ISDN user network interface layer 3 specification for basic call control. When a Gateway is implemented between an IP infrastructure and the public switched telephone network **1200**, the H.225 call signaling may be converted to the appropriate Q.931 signaling when a PRI ISDN digital interface is used. Such an implementation may use T1 CAS signaling to interface with PSTN as per the T1 specification (AT & T PUB 43801).

When an endpoint (a first user) initiates a call, call setup takes place using the call control messages defined in H.225. A receiving endpoint sends an Alerting message to indicate that the called party (a second user) has been alerted of an incoming call. In the case of a Gateway, the receiving endpoint sends the Alerting message when it receives a ring indication from the PSTN. In accordance with the current H.323 specification, if the endpoint can respond to a Setup message with a Connect, Call Proceeding, or Release Complete within 4 seconds, it need not required to send the Alerting message. An end-point sending the Setup message can expect to receive either an Alerting, Connect, Call Proceeding, or Release Complete message within 4 seconds after successful transmission.

With reference to **Fig. 3**, the VoIP manager **3070** receives the Call Setup Message from a Transport Service Access Point. In some embodiments, the Transport Service Access Point is to the internet-type network **1100** via a dedicated Frame Relay

connection. The IP packets may be encapsulated in the Frame Relay packet according to RFC 1490. This then goes through the data protocol stack and becomes an IP packet, i.e. a Call Setup Message to the VoIP manager **3070**. The VoIP manager **3070** may then interact with the call management module **3110** to determine if there is a POT line available for this incoming call. If there is a line available for the incoming call, the VoIP manager **3070** sends a Call Proceeding message to the client endpoint.

Fig. 8A depicts a flow diagram for a method for communicating an AudioTM message to a VoIP client or Etherphone in accordance with an illustrative embodiment. Process flow initiates at an 'start' terminal **8100** in response to an incoming call signal from the first VoIP client **1530** being detected by the gateway. Call establishment occurs as described above.

Next, a 'send AudioTM to first user' decision process **8200** determines if an AudioTM is to be sent to the first user. If not, the 'send AudioTM to first user' decision process **8200** exits through its 'no' branch and process flow completes through an 'end' terminal **8500**.

If an AudioTM is to be sent, process flow continues to the 'send Call Accepted' process **8300** sends a signal from the gateway to the first VoIP client **1530** to indicate that the call signal has been accepted. The signal may be, for instance, a Connect Message according to the H.225 protocol. Processing continues to an 'AudioTM communication' process **8400**. The 'AudioTM communication' process **8400** then executes in a fashion analogous to the 'AudioTM communication' process **4700** or the 'feedback AudioTM communication' process **4600** of **Fig. 4**. Processing completes through an 'end' terminal **8500**.

Incoming communication handling will now be described. For purposes of illustration of incoming communication handling a second user operating the second VoIP client **1630** is assumed to be receiving an incoming communication from a first user operating one of the first set of aural communication devices **1500**.

Fig. 8B depicts a flow diagram for a method for communicating an AudioICM message to a VoIP client or Etherphone in accordance with an illustrative embodiment. Process flow initiates at a 'start' terminal **8600** and continues to a 'send AudioICM to second user' decision process **8700** that determines if an AudioICM should be sent to the second user. If not, the 'send AudioICM to second user' decision process **8700** exits through its 'no' branch and processing completes through an 'end' terminal **8950**.

If an AudioICM should be sent, the 'send AudioICM to second user' decision process **8700** exits through its 'yes' branch to a ''Call Accepted' received' decision process **8800**. The ''Call Accepted' received' decision process **8800** determines if a 'Call Accepted' signal has been received from the second VoIP client **1630**. When the 'Call Accepted' signal is received, the ''Call Accepted' received' decision process **8800** exits through is 'yes' branch and process flow continues to an 'AudioICM communication' process **8900**. The 'AudioICM communication' process **8900** then executes in a fashion analogous to the AudioICM communication method **7000** of **Fig. 7**. Processing completes through an 'end' terminal **8500**.

2.B. GATEWAY AT CENTRAL OFFICE

In some embodiments, the VoIP gateway is not located at a customer's premises and may be located, for instance, at a central office. Reference may be made to **Fig. 1C** for an exemplary network topology in which such embodiments may be deployed. In accordance with such embodiments, the aural communication device may be, for instance, a computer programmed to provide a VoIP client such as the second VoIP programmed computer **1680**, an IP phone such as the VoIP phone **1665**, or a conventional phone connected through a converter and working as an IP phone, such as the POTS phone **1670** and the VoIP converter **1675**.

Generally, when the VoIP gateway is at a central office, practicing versions of the invention is similar to that described above in connection with VoIP gateways at

customer premises. Differences may be illustrated in connection with the second VoIP programmed computer **1680** as an illustrative embodiment.

The second VoIP programmed computer **1680** may be a conventional PC running the WINDOWS operating system from Microsoft and NetMeeting client software. The second VoIP programmed computer **1680** may have a built-in modem (analog or digital) for access to the internet-type network **1100**. The second VoIP programmed computer **1680** may also operate with a conventional phone going through a converter and then connected to the second VoIP programmed computer **1680** through, for instance, an Ethernet interface, Universal Serial Bus, IEEE 1394 serial bus, PCI bus, or other suitable means. Speakers and a microphone are operatively coupled with the second VoIP programmed computer **1680** to act as the ear and mouth pieces of a phone. The NetMeeting software converts audio and voice into H.323 compliant IP format. This information is communicated through the modem and to the second DSLAM **1875** or a Remote Access System ("RAS"), e.g. a conventional modem bank or Cable Head-end system (not shown). The second DSLAM **1875**, RAS, or Cable Head-end communicates the information to the second VoIP gateway **1825** that may follow the H.323 protocol to determine routing. If routing to the public switched telephone network **1200** is needed, the second VoIP gateway **1825** performs necessary conversion between VoIP signaling and PSTN voice.

In this situation, programmed instructions may configure the second VoIP gateway **1825** to embody aspects of, and perform methods according to, versions of the invention illustrated and described herein. Typically this may involve, at least, providing capability to transfer messages to and from message servers and local storage for messages retrieved from message servers. AudioMT, AudioTM, and AudioICM transmission may be provided as described above with reference to the processes in **Fig. 4–Fig. 7** by suitably modifying the call management module **3110** as one of skill in the art will readily appreciate having the benefit of this disclosure.

However, when a simulated dial tone is not provided by the second VoIP gateway **1825**, AudioMT is not supported. AudioTM and AudioICM transmission are provided as described above in connection with embodiments where the VoIP gateway is located on a customer's premises.

In some embodiments, the invention may operate with speaker verification system. The speaker verification system may include a phone control interactive voice response ("IVR") system and a voice/key server. One illustrative embodiment may be understood with reference to **Fig. 1E** where IVR systems are shown on the side of the VoIP gateways of **Fig. 1C** (omitted in **Fig. 1E** for clarity). Shown in **Fig. 1E** are a first phone/key IVR **1250**, a second phone/key IVR **1260**, a first voice/key server **1150**, a second voice/key server **1160**. Suitable structures for these systems are commercially available and the PhoneKey system from ITT Industries of San Diego, California and its partner Buytel of Dublin, Ireland, may be used. The first phone/key IVR **1250** communicates with the first central office switch **1220**, the first message switch **1270**, and the first voice/key server **1150**. The second phone/key IVR **1260** analogously communicates with the second central office switch **1230**, the second message switch **1280**, and the second voice/key server **1160**. In operation, when a user of, for instance the VoIP phone **1665**, initiates a call, the second central office switch **1230** will route the call to the second phone/key IVR **1260** for speaker verification based on the subscription profile for the line associated with the VoIP phone **1665**. The second phone/key IVR **1260** prompts the user to verify his or her identification by speaking a particular phrase, e.g., the user's name, a personal identification number, or other suitable phrase. The second phone/key IVR **1260** then consults subscriber records accessible through the second voice/key server **1270** to determine if the spoken particular phrase can be verified as associated with a particular subscriber, i.e. the "voice print" matches. If so, validation information is returned from the second phone/key IVR **1260** to the second central office switch **1230** and the call continues as if it were originated from a phone connected to the public switched telephone network **1200**. In alternative embodiments, a user may

prompted to identify themselves by entering an identifier such as personal identification number ("PIN") rather than with an IVR system.

As described above with reference to **Fig. 1E**, the second phone/key IVR **1260**, for instance, is shown connected to the second central office switch **1230**, i.e., on the PSTN side of the VoIP gateway. Alternatively, VoIP gateways, as shown in **Fig. 1C**, may be configured to detect if a caller is initiating a call from a line associated with a particular class of subscribers. The VoIP gateway could then route the call to a particular location in the public switched telephone network **1200**; thereafter the user's identification could be verified using one of the above-described means.

Whatever the means of user identification verification, once the user's identification is verified, additional features of the invention may be conveniently deployed. For instance, a more personalized messaging experience may be provided to the user by selection of informational messages responsive to demographic, psychographic, or socio-economic information in the user's subscription profile. Further, user identification verification facilitates the consummation of voice-based commercial transactions as the known user's identification more reliably secures payment of a price. In some embodiments a feedback message may prompt a user to enter into a commercial transaction, the user's identification could be verified using means described above, and the price the user agrees to pay in the commercial transaction could be added to the user's periodic bill from the service provider. One of skill in the art will appreciate that these features of the invention are not limited to embodiments using VoIP communications.

3. *AURAL COMMUNICATION DEVICE CONNECTED TO PBX OR KEY TELEPHONE SYSTEM BEFORE CONNECTED TO CPE*

In some embodiments of the invention one of the first set of aural communication devices **1500** is connected to a PBX or Key Telephone System before it is connected to the first CPE device **1400** (through a gateway). In these embodiments the first CPE device **1400** does not have direct control over the first POTS phone **1510** and accordingly does not support transmission of AudioMT messages. Other features of previously described embodiments may still operate when the first POTS phone **1510** is connected to a PBX or Key Telephone System before it is connected to the first CPE device **1400**.

In some embodiments when the first POTS phone **1510** is connected to a PBX or Key Telephone System, the signaling format may need to be modified. For an analog trunk type connection, E&M signaling format is suitable. In the case of a digital trunk connection, suitable signaling formats include, for example, T1 (or E1) running ISDN PRI and an appropriate signaling format supported by the PBX; in the case of T1, this may be CAS.

4. *PHONE DIRECTLY CONNECTED TO THE CENTRAL OFFICE SWITCH*

Some embodiments of the invention involve a phone, such as the third POTS phone **1710** directly connected to a central office switch, such as the first central office switch **1220**.

In some embodiments software operating the central office switch is modified to implement the functions of the call management module **3110**, the message server agent **3030**, and the message manager **3040** as has been previously described. In

addition, a network connection between the central office switch and a message server is established. The message server may be on a public network such as the internet or may be on a private network. The precise location of the message server in a network topology is not fundamental.

In a modern PSTN network, a typical network architecture is shown as in **Fig. 1B**. The public switched telephone network **1200** may rely on SS7 as the basic infrastructure needed for a service switching point **1210** ("SSP"), which provides the local access as well as an ISDN interface for a signaling transfer point **1220** ("STP"). The signaling transfer point **1220** provides packet switching of message-based signaling protocols for use in the network and for a service control point **1230** ("SCP"), which provides access to a network database (not shown). The service control point **1230** communicates with a service management system **1240** ("SMS"), that provides a human interface to the database, as well as the capability to update the database when needed. An intelligent peripheral **1250** may also be added. The intelligent peripheral **1250** may provide resource management of devices such as voice response units, voice announcers, and DTMF sensors for caller-activated services.

An additional feature of a modern PSTN is service-independent Advanced Intelligent Networks (AIN) capabilities. With AIN capabilities, service providers are able to load service logic in a SCP. The service logic typically allows for triggering capabilities to be programmed in order to route the calls of the service's subscribers. Many different triggering conditions can be programmed including, for instance, an off hook' condition or a number of digits collected. Illustrative versions of the invention may operate with systems that route calls in this fashion.

In one illustrative version, the triggering condition could be an 'off hook' condition, e.g, a subscriber has taken his or her aural communication device off hook. In response to this triggering condition, calls from this subscriber's line could be

routed to a particular switch that performs functions analogous to those described above in connection with **Fig. 2** through **Fig. 7**. In this illustrative version, when a subscriber takes his or her phone off hook to place a call, the off-hook condition is detected and this triggers the routing of the call to the particular switch. The subscriber may then be transmitted informational message(s) under the control of the particular switch in a fashion analogous to that described above in connection with **Fig. 2** through **Fig. 7**. In another version, the service providers could trigger on a single digit key such as '* ' or '#' to offer the service of similar functionality. In still others, another provider could trigger on multiple digits for the service.

Another illustrative embodiment would be where the central office switch is programmed to route calls originating from subscribers to the device by appending a routing code to the calls. The appended routing code triggers the routing of calls to the device. One example of a suitable routing code are conventional Primary Interexchange Carrier ("PIC") codes, although one of skill in the art will appreciate others.

In one version of this illustrative embodiment, the central office switch automatically appends the PIC code. Conventionally, central office switches use routing codes, such as PIC codes, to route long distance calls based on the subscriber's predetermined long-distance carrier. In this version, a subscription service could be established. The subscription service could offer, for instance, subsidized telecommunication products or services in exchange for the subscriber's receiving informational messages from a sponsor. A central office switch could be configured to append an appropriate PIC code associated with the service as the default routing code for subscribers in a fashion analogous to default routing codes for long-distance calls. Calls from subscribers to the service could then be routed to a particular switch which has been configured to implement the functions for and features of transmission of informational messages as described in this specification.

In another version, the routing code is entered by the user of the aural communication device. Conventionally, central office switches also use routing codes entered by the user, to route calls for particular long distance carriers. For instance, Telecom*USA of Arlington, Virginia, offers a service commercially available in the United States by which a user dialing "10 10 321", will have their call routed to Telecom*USA for long-distance calling. This conventional method of routing calls could be used to route calls to the particular switch which has been configured to implement the functions for and features of transmission of informational messages as described in this specification.

In still another version, the operator assistance code could be entered by the user of the aural communication device. This assistance code could be used by the central office switches conventionally to route calls from the caller to the particular switch which has been configured to implement the functions for and features of transmission of informational messages as described in this specification.

The aforementioned embodiments in which calls are routed to a particular switch provide a convenient architecture for commercial adoption of versions of the invention: the particular switch practicing features of the invention may be acquired by the operator of a central office switch (the particular switch may be collocated with existing central switching equipment or remotely disposed therefrom), existing routing methods may be used to then route subscriber's calls to the particular switch, and the particular switch may provide informational messages to subscribers. In the aforementioned embodiments, the functions described above in connection with **Fig. 2** through **Fig. 7** may be implemented. They may be implemented, for instance through software that configures the central office switch to implement the functions. Alternatively, the particular switch may be an IVR system designed to perform functions analogous to those described above in connection with **Fig. 2** through **Fig. 7**. However, as one of skill in the art will appreciate, the call density is typically high in this type of embodiment. Accordingly, suitable interfaces should be used. Some

embodiments could use multiple T1/E1, others could use DS3; others could use still higher bandwidth interfaces.

In some variations of the aforementioned embodiments, an IVR system could be employed for verification of a subscriber. Again making reference to **Fig. 1E**, a first message switch **1270** and a second message switch **1280** will be noted. Referring to the first message switch **1270** as also illustrative of the second message switch **1280**, it can be appreciated that the first message switch **1270** may be flexibly integrated with existing central switching hardware such as the first central office switch **1220** that could route calls to the first message switch **1270**, i.e., the first message switch **1270** may be the particular switch described above. When a user of, for instance the third POTS phone **1710**, initiates a call, the call could be routed by the first central office switch **1220** to the first phone/key IVR **1250** for user verification. The user may verify himself or herself by voice print matching in conjunction with the first phone/key IVR **1250**, by entering personal identification information from a phone keypad, or through other means. If the voice print matches or the personal identification information validates, the system notifies the first message switch **1270** of the caller's identification and a voice channel on which the call is to be carried out. The first message switch **1270** could then consult the user's subscription profile to determine whether one or more informational messages should be sent transmitted to the user. As this illustrative example assumes the user is initiating a call, the informational messages would be AudioMT or AudioTM (with or without feedback options). The first message switch **1270** (configured for communication with the first message server **1110**) can then transmit one or more informational messages retrieved from the first message server **1110** to the user. After the transmission of the informational message(s), the user's call proceeds (in the case of an AudioMT(s), an additional dial tone may be provided to the user after the transmission of the informational message(s)).

In still other embodiments an intelligent peripheral such as the intelligent peripheral **1250** may be used to implement features of the invention. Connection to a message server such as the first message server **1110** is made with a digital facility such as that typically found in a central office. Typically, a central office switch, such as the first central office switch **1220** has a connection to a digital facility. The digital facility allows signaling information between central office switches to be consolidated and sent through its own network apart from the voice network. This method is used in ISDN and SS7 today. In addition, this method of signaling is capable of sending and receiving messages, including, for instance, AudioTM, AudioICM including information retrieved from a remote database.

As will be appreciated by one of skill in the art having the benefit of this disclosure, a connection to, for instance, the first message server **1110** may be obtained by having the first message server **1110** act as an intelligent peripheral such as the intelligent peripheral **1250**. The first message server **1110** would provide support for a protocol appropriate for communication with the central office switch (currently, SS7 could be used). The first message server **1110** may then either be directly connected to the central office switch through digital connection or through the service control point **1230**. Software running on a central office switch or a SCP may then be modified to implement functions according to the invention as described herein.

For instance, when a central office switch detects that the subscriber's aural communication device is off-hook, the switch could be configured by programmed instructions to assemble the dialing digits and communicate them to the first message server **1110** and to communicate the subscriber's line information to the intelligent peripheral **1250**. If an AudioMT, an AudioTM, or an AudioICM should be sent to the subscriber, it will retrieve the message from the message server Intelligent Peripheral, and then send it to the subscriber.

In addition, all or portions of a wireless communications network may also be integrated with the public switched telephone network **1200**. When a wireless network is integrated with the public switched telephone network **1200**, users of wireless mobile units may be transmitted informational messages as has been described above with reference to a phone directly connected to a central office switch. Other versions of the invention operate with other configurations of a wireless network as illustrated below.

5. WIRELESS

As noted, some embodiments of the invention operate with wireless networks. Illustrative embodiments are described below in the context of a cellular wireless network with the understanding that one of skill in the art will comprehend principles of the invention in this description applicable in other types of wireless networks (also within the scope of the invention).

5.A OUTGOING COMMUNICATION

Fig. 9 depicts a flow diagram of a method for communicating an AudioTM in accordance with an illustrative cellular embodiment. Process flow of a 'cellular AudioMT communication' method **9000** initiates at a 'start' terminal **9010** and continues to a 'call initiation' data block **9020**. The 'call initiation' data block **9020** is received by the base transceiver stations **1903** and comprises an identification of the mobile unit **1905** and a serial number. A 'phone ID transmission' process **9030** passes this information to the mobile switching center **1920** and a 'calling phone authentication' process **9040** validates the calling phone number with the home local register **1930** and the visitor location register **1950** and authenticates the identity of the mobile unit **1905**.

Process flow continues to a 'message server consultation' process **9050**. The 'message server consultation' process **9050** is performed by the mobile switching

center **1920** and includes the mobile switching center **1920** transmitting mobile unit **1905** subscriber identification information to the message server **1990**. The message server **1990** executes an 'AudioTM to cellular' decision process **9060** to determine whether AudioTM should be sent to the mobile unit **1905** at this time.

If the AudioTM should not be sent, the 'AudioTM to cellular' decision process **9060** exits through its 'no' branch and process flow continues to a 'call placement' process **9240** that initiates a conventional call connection sequence performed by the mobile switching center **1920**.

If the AudioTM should be sent, the 'AudioTM to cellular' decision process **9060** exits through its 'yes' branch and process flow continues to an 'an AudioTM retrieval' process **9070** that retrieves the AudioTM from the message server **1990** for sending to the mobile unit **1905**. Process flow continues to a 'feedback AudioTM' decision process **9080** that consults the AudioTM to determine whether the AudioTM will offer feedback options. If not, an 'AudioTM transmission' process **9090** transmits the AudioTM to the mobile unit **1905**, a 'messages sent flag setting' process **9230** sets a flag indicating that the AudioTM was sent and process flow continues to the 'call placement' process **9240**.

If the AudioTM, will offer feedback options, the 'feedback AudioTM' decision process **9080** exits through its 'yes' branch and process flow continues to an 'feedback AudioTM transmission' process **9100** that transmits the feedback AudioTM to the mobile unit **1905**. The message may offer any of the feedback options previously discussed. An 'AudioTM feedback indication' decision process **9110** awaits an indication from the mobile unit **1905** of the selection of a feedback option. If no indication of selection of a feedback option is received within a suitable predetermined time limit, e.g., 5 seconds, the 'AudioTM feedback indication' decision process **9110** exits through its 'no' branch and process flow continues to the 'messages sent flag setting' process **9230**.

If an indication of the selection of a feedback option is received, the 'AudioTM feedback indication' decision process **9110** exits through its 'yes' branch and process flow continues to a 'feedback indication storage' process **9120** that posts the selected feedback option to the message server **1990**. Next a 'call redirection' decision process **9130** determines if the particular selected feedback option involves redirecting the user of the mobile unit **1905** to complete a call to another entity before completion of the call from the 'call initiation' data block **9020**. If so, the 'call redirection' decision process **9130** exits through its 'yes' branch and a 'number storage' process **9140** stores the number from the 'call initiation' data block **9020** while a 'redirected call' process **9150** executes. The 'redirected call' process **9150** involves the mobile switching center **1920** consulting the message server **1990** for the number to which a call should be placed given the particular feedback selection. In some embodiments, a user of the mobile unit **1905** may use an alias dialing feature as described above (in connection with the 'feedback AudioMT communication' process **4300**) and below (in connection with **Fig. 16**) and the message server **1990** may map the dialed alias to the appropriate number to call. During the 'redirected call' process **9150** the user of the mobile unit **1905** may carry on a conventional phone conversation with the entity associated with the user's feedback option selection. From the 'redirected call' process **9150** process flow continues to a 'number retrieval' process **9160** in which the mobile switching center **1920** retrieves the originally-called number from the 'call initiation' data block **9020** that was stored by the 'number storage' process **9140**.

If the particular selected feedback option does not involve redirecting the user of the mobile unit **1905** to complete a call to another entity, the 'call redirection' decision process **9130** exits through its 'no' branch and process flow continues to an 'automated output' decision process **9170** that determines if an automated output should be sent in response to the feedback selection. If so, the 'automated output' decision process **9170** exits through its 'yes' branch and process flow continues to an 'output generation' process **9180** that consults the message server **1990** for the

particular type of automated output that should be generated. Any of the types of automated output discussed above may be generated. Process flow continues to an 'output communication' process **9190** that communicates the automated output to a suitable receiving entity.

From the 'output communication' process **9190** or the 'number retrieval' process **9160** process flow continues to a 'remaining feedback' decision process **9200** that determines if the user has indicated an additional feedback option selection. The user may be presented with such an option by the 'output communication' process **9190** prompting them with an audio message of additional feedback options or the AudioTM retrieved by the 'an AudioTM retrieval' process **9070** may indicate that any of a menu of feedback options could be selected at this point of the process flow. If the 'remaining feedback' decision process **9200** detects that an additional feedback option has been selected, it exits through its 'yes' branch and a 'next feedback message selection' process **9210** retrieves the appropriate next message. Process flow continues to a 'next feedback message communication' process **9220** that communicates the next message to the mobile unit **1905** and process flow returns to the 'feedback indication storage' process **9120** for the next iteration.

If the 'remaining feedback' decision process **9200** does not detect that an additional feedback option has been selected, it exits through its 'no' branch and process flow continues to the 'messages sent flag setting' process **9230** and the 'call placement' process **9240**. Processing completes through an 'end' terminal **9250**.

5.B INCOMING COMMUNICATION

Fig. 10 depicts a flow diagram of a method for communicating a TextICM message in accordance with an illustrative embodiment. The 'TextICM communication' method **10000** may operate with elements of a suitable cellular network topology including, for instance, the exemplary cellular network architecture **1900** discussed above in connection with **Fig. 1D**. Process flow initiates at a 'start'

terminal **10100** when a user is placing a call to the mobile unit **1905**. An 'incoming call' input block **10200** comprising the identification and serial number of the mobile unit **1905** as well as the called number is communicated to the mobile switching center **1920** that validates that called number with suitable lookup in the home local register **1930** and the visitor location register **1950**. Process flow continues to a 'TextICM' decision process **10300** that includes the mobile switching center **1920** providing a query to the message server **1990** to determine whether a TextICM should be communicated to the mobile unit **1905**. The query to the message server **1990** may include, for instance, identification information from the mobile unit **1905** such as the serial number, and the identification information may be used by the message server **1990** in conjunction with profile information to generate or select a TextICM responsive to the particular identification information. As one of skill in the art will appreciate, demographic, pshychographic, and profile information may be used in generating or selecting the TextICM.

If the TextICM should not be sent the 'TextICM' decision process **10300** exits through its 'no' branch and process flow continues to an 'incoming call handling' process **10800** that completes the placement of the incoming call to the mobile unit **1905**. If the TextICM should be sent to the mobile unit **1905**, the 'TextICM' decision process **10300** exits through its 'yes' branch and process flow continues to a 'TextICM retrieval' process **10400** in which the TextICM is retrieved from the message server **1990**. Process flow continues to a 'text connection establishment' process **10500** in which the mobile switching center **1920** establishes a data communication connection to the mobile unit **1905**. Next, a 'TextICM transmission' process **10600** transmits the TextICM to the mobile unit **1905** where it may be perceived by the user thereof. A 'text connection termination' process **10700** terminates the text connection and process flow continues to the 'incoming call handling' process **10800** where the incoming call may be conventionally completed with the mobile unit **1905**. Process flow completes through an 'end' terminal **10900**.

User verification, as described above with reference to **Fig. 1E**, could also be employed with cellular embodiments. To employ speaker verification in a cellular embodiment, a speaker verification system, such as the PhoneKey system from ITT Industries and Buytel, can be installed between the mobile switching center **1920** and the public switched telephone network **1200**. The process logic would then be similar to that described above with reference to the third POTS phone **1710**.

It will be understood by one of skill in the art having the benefit of this disclosure that the principles of the invention are not limited to messages of the type refereed to above as a TextICM. Rather, this type of message has been used as the basis of this description as it what the current generation of wireless communication devices in broad distribution can support. It is contemplated that as wireless communication device technology evolves it will support, for instance video and other media. It is contemplated that one of skill in the art having the benefit of this disclosure will be able to apply the principles of the invention to operate with such devices, and such application is contemplated as within the scope and spirit of the present invention.

6. MULTIMEDIA MESSAGES

Some embodiments of the invention involve the transmission of non-audio or multimedia messages.

Suitable systems for such embodiments include, for example, a programmed general-purpose computer, or special-purpose computing machinery, that may be situated proximate to a conventional telephone. An office or home environment are typical situations where suitable systems may be found. The computer may be used to provide an enhanced message transmission system.

In an illustrative embodiment, a personal computer is situated near a phone, for example the first coupled PC & phone **1520**. The phone and the computer are

connected with an IAD such as the first CPE device **1400**. The computer is connected to the IAD via a LAN such as the first LAN **1540**. The IAD has a hardware and software architecture as described in connection with **Fig. 2** and **Fig. 3** with additional logic for sending a signal ("SIGicm") to the computer via the LAN when the IAD receives an indication that there is an incoming phone call for the phone line associated with the computer.

An IAD application runs on the computer that enhances the types of messages that may be transmitted in conjunction with use of the phone. In the illustrative embodiment, the IAD application provides conventional computer telephony functions including, for example, name based dialing, voice recognition dialing, multiple call forwarding and screening functions, etc., although the presence of these functions are not fundamental to the invention and may be omitted in alternative embodiments. The computer may also comprise a microphone and speakers that can operate as the mouthpiece and earpiece of a conventional telephone so that the computer may be used to send and receive telephone calls. The IAD application includes a driver to detect the SIGicm from the IAD. Further, the IAD application can generate pop-up-type windows or banners, and play multimedia, audio, or non-audio messages under the control of the IAD.

Upon installation of the IAD application, the IAD application sends a message to the IAD indicating that the computer is running the IAD application. The IAD then extracts a network address, such as an IP address, application software version, relevant host hardware and software configuration information, etc. from the computer. Then, the IAD sends informational messages in audio, non-audio, or multimedia form for storage on the computer.

Operation of the computer, phone, and IAD to provide informational messages in connection with outgoing communication will now be described. For this

discussion it is assumed that a first user of a first PC & phone is making an outgoing call.

Fig. 11 depicts a flow diagram of a method for communicating informational messages to a PC & phone in conjunction with an outgoing call **11000** in accordance with an illustrative embodiment. Process flow initiates at a 'start' terminal **11010** and receives an indication a first PC & phone is 'off hook" **11020**. This may result from the first user picking up the handset of the phone or initiating a call though the computer. The 'off hook' condition is detected by an ''off hook' detection' process **11030** and process flow continues to an 'AudioMT to first user' decision process **11050** that determines if the first user should receive an AudioMT message. If so, the 'AudioMT to first user' decision process **11050** exits through its 'yes' branch and process flow continues to a 'AudioMT communication' process **11060** that communicates the AudioMT for perception by the first user. From the 'AudioMT communication' process **11060**, or if the 'AudioMT to first user' decision process **11050** exits through its 'no' branch, process next flows to a 'message to computer display' decision process **11070** that determines whether a PopupM should be sent to the computer display.

If the PopupM should not be sent to the computer display, process flow continues to a 'dial tone to first user' process **11090**. If the PopupM should be sent to the computer display, the 'message to computer display' decision process **11070** exits through its 'yes' branch and a 'send message to display' process **11080** sends a signal to the IAD application running on the computer to present the PopupM on the computer display. Process flow then continues to the 'dial tone to first user' process **11090** which provides a dial tone to the aural communication device of the first user and an outgoing call handling' process **11100** that accepts dialing digits from the first user's aural communication device and initiates the outgoing call.

Process flow continues to an 'AudioTM to first user' decision process **11130**. The 'AudioTM to first user' decision process **11130** determines if an AudioTM should be sent to the first user. If the AudioTM should not be sent to the first user process flow continues to a 'call establishment and handling' process **11150** that completes establishing the first user's call. If the AudioTM should be sent to the first user a 'AudioTM communication' process **11140** executes in a manner analogous to that described in connection with **Fig. 6A** and process flow continues through the 'call establishment and handling' process **11150** and completes through the 'end' terminal **11120**.

Operation of the computer, phone, and IAD to provide informational messages in connection with incoming communication will now be described. For this discussion it is assumed that a second user of a second PC & phone is receiving an incoming call from a first user of a first aural communication device.

Fig. 12 depicts a method for communicating informational messages to a PC & phone in conjunction with an incoming call **12000**. Process flow initiates at a 'start' terminal **12010** and continues to receive an 'incoming call' data block **12020** indicating that an incoming call has been received. Process flow continues to a 'ring and ring back tone' process **12030** that sends a ring signal to the second PC & phone and a ring back tone to the first aural communication device. Next, a 'display message' decision process **12040** determines if a PopupM should be sent to the computer display. If a message should be sent to the computer display, the 'display message' decision process **12040** exits through its 'yes' branch and an 'message display' process **12050** sends the PopupM to the IAD application running on the computer to initiate the display of the message. From the 'message display' process **12050**, or if the 'display message' decision process **12040** exits through its 'no' branch, process flow continues to an 'answered via PC' decision process **12060**.

The 'answered via PC' decision process **12060** determines if the incoming call has been answered with the computer. If so, process flow continues to a 'set 'Answered by PC' flag' process **12070** that sets a flag indicating the incoming call has been answered with the computer and process flow completes through an 'end' terminal **12080**.

If the incoming call was not answered with the computer, process flow continues to an 'answered by handset' decision process **12090**. The 'answered by handset' decision process **12090** determines if the incoming call has been answered with the phone handset. If the incoming call was not answered with the phone handset, process flow continues to a 'ring limit' decision process **12100**. The 'ring limit' decision process **12100** determines if a limit on the number of rings has been exceeded. If the ring limit has been exceeded, process flow continues to an 'set 'No Answer' flag' process **12110** that sets a flag indicating that the incoming call was not answered and process flow completes through the 'end' terminal **12080**.

If the incoming call was answered with the phone handset, the 'answered by handset' decision process **12090** exits through its 'yes' branch to enter an 'AudioICM to second user' decision process **12120**. If an AudioICM is not to be transmitted, the 'AudioICM to second user' decision process **12120** exits through its 'no' branch and process flow completes through the 'end' terminal **12080**.

If an AudioICM is to be transmitted, the 'AudioICM to second user' decision process **12120** exits through its 'yes' branch and a 'retrieve AudioICM' process **12130** retrieves the AudioICM from the memory subsystem **2200**. Next, if the AudioICM is compressed, a 'compressed message' decision process **12140** exits through its 'yes' branch and a 'decompression' process **12150** decompresses the AudioICM. Process flow continues to a 'text to audio' decision process **12160** that exits through its 'yes' branch if the AudioICM is in text form and should be converted to audio. A 'text to audio conversion' process **12170** performs the conversion. Process flow continues to

a 'send AudioICM to second user' process **12180** that transmits the AudioICM in audio form to the second phone for perception by the second user.

Process flow continues to a 'remaining messages' decision process **12190**. If no messages remain for transmission, the 'remaining messages' decision process **12190** exits through its 'no' branch and a 'set 'Message Sent' flag' process **12210** sets a flag indicating that the AudioICM was sent and process flow completes through the 'end' terminal **12080**. If there is a remaining message for transmission, the 'remaining messages' decision process **12190** exits through its 'yes' branch and process flow continues to a 'continue ring back tone' process **12200**. As was described above in connection with the 'continue Ring Back Tone' process **7160** of **Fig.** 7, the 'continue ring back tone' process **12200** of **Fig. 12** may provide the Ring Back Tone to the originator of the incoming communication. In other versions, the 'continue ring back tone' process **12200** sends a call connected message to the switching system to indicate that the incoming communication has been accepted and thereafter a message may be sent to the first user indicating the first user that his or her call is connecting with the second user. In other versions the message indicating that his or her call is connecting with the second user may be sent before the call connected message is sent to the switching system. In either case, the message indicating that his or her call is connecting with the second user may be substituted by, or accompanied with, an informational message such as has been described above. Process flow returns to the 'retrieve AudioICM' process **12130** to retrieve the remaining message and begin the next iteration.

6.A. VOICE OVER IP CLIENT RUNNING ON A PC CONNECTED TO AN IAD

In some embodiments of the invention where the computer is running the IAD application, the computer is also running an application that allows the computer to act as a VoIP client.

Fig. 13A depicts a flow diagram for a method for communicating a PopupM message to a VoIP client in conjunction with an outgoing call in accordance with an illustrative embodiment. Process flow initiates at an 'start' terminal **13010** and continues to a 'VoIP call initiated' data block **13020** that is detected by the IAD and indicates that a VoIP call has been initiated by the VoIP client running on the computer. Process flow continues to a 'send PopupM' decision process **13030** that determines whether a PopupM should be sent. If the PopupM should be sent, process flow continues to a 'signal PC' process **13040** where the IAD communicates a signal to the IAD application running on the computer to display the PopupM.

From the 'signal PC' process **13040**, or if the 'send PopupM' decision process **13030** exits through its 'no' branch, process flow continues to a 'continue VoIP call' process **13050** that continues the VoIP conventionally. Process flow completes through an 'end' terminal **13060**.

Fig. 13B depicts a flow diagram for a method for communicating a PopupM message to a VoIP client in conjunction with an incoming call in accordance with an illustrative embodiment. Process flow initiates at a 'start' terminal **13070** and continues to a 'VoIP call received' data block **13080** that is detected by the IAD and indicates that an incoming VoIP call has been received. Process flow continues to a 'send PopupM' decision process **13090** that determines whether a PopupM should be sent. If the PopupM should be sent, process flow continues to a 'signal PC' process **13100** where the IAD communicates a signal to the IAD application running on the computer to display the PopupM.

From the 'signal PC' process **13100**, or if the 'send PopupM' decision process **13090** exits through its 'no' branch process flow continues to an 'incoming call indication' process **13110** where the IAD passes a Call Setup message to the VoIP client running on the computer. Process flow continues to a 'continue VoIP call' process **13120** that continues the VoIP conventionally. Process flow completes through an 'end' terminal **13130**.

6.B. FEEDBACK MULTIMEDIA MESSAGES

Some embodiments provide feedback multimedia messages. In **Fig. 14** and **Fig. 15** described below, to illustrate structures performing functions in accordance with an illustrative version of the invention, headings are provided in the figures indicating the structures performing the processes below. While described below in the context of an "HTTP Server" and "IAD/PBX" it will be understood that these structures are illustrative and not limiting. Further a "Computer/Phone" as used below refers to a device, such as an integrated computer and telephone, capable of performing ordinary telephonic communications and the function of a general purpose programmable computer.

Fig. 14 depicts a flow diagram for a method for communicating a feedback PopupM message to a computer/phone in conjunction with an outgoing communication in accordance with an illustrative embodiment. An 'outgoing feedback PopupM' method **14000** initiates at a 'start' terminal **14010** and process flow continues to receive an 'off hook' input block **14020** indicating that the computer/phone has detected that a user desires to make a call. The 'off hook' input block **14020** is passed to the IAD/PBX and received by a 'detects off hook' process **14030**.

Process flow continues in the IAD/PBX to a 'send PopupM' decision process **14040** that determines whether a PopupM should be sent to the computer/phone. The 'send PopupM' decision process **14040** may consult a profile in local or remote data store as has been previously illustrated.

If the PopupM should be sent, the 'send PopupM' decision process **14040** exits though its 'yes' branch and process flow continues to a 'PopupM communication' process **14050** where the IAD/PBX retrieves the PopupM and

transmits it to the computer/phone where a 'PopupM display' process **14060** presents it to the user.

Some embodiments allow feedback to take place with a browser application running on the computer/phone. Others allow feedback to take place with a call being placed to a third party. The 'PopupM display' process **14060** may prompt the user for either of the previously-identified types of feedback. For instance, the PopupM could provide a banner that comprises hyperlinks that, when selected by the user, are followed by a browser application. The hyperlinks may be to a sponsor's web site or, more generally, to any network resource addressable in the internet-type network **1100**. Alternatively the PopupM could have a sound file prompt the user to make such a selection. Still further, the PopupM may use a banner or sound file to prompt the user to select an item that will result in a call being placed to a third party, for instance, a sponsor, other merchant, etc. Both the Uniform Resource Identifier in the hyperlink and the phone number of the third party to be called are examples of resource identifiers. More generally, other embodiments of the invention may use other resource identifiers.

From the 'PopupM display' process **14060** process flow continues to a 'resource identifier selection typing' process **14070** that determines a type of resource selected by the user. If the user does not select a feedback option within a predetermined time, process flow continues to an 'outgoing call handling' process **14120** in which the IAD/PBX accepts a dialing sequence and establishes the user's outgoing call.

If the resource identifier selected by the user is a phone number of a third party to be called, the 'resource identifier selection typing' process **14070** passes this information to the IAD/PBX and a 'RI call handling' process **14140** executes in which the user completes the call to the third party. Process flow then continues to the 'outgoing call handling' process **14120**.

If the resource identifier selected by the user is Uniform Resource Identifier, the 'resource identifier selection typing' process **14070** passes this information to a browser application running on the computer/phone which sends a request message for the identified resource. A 'uniform resource identifier request' input block **14090** is received by the server from which the identified resource is available and process flow continues to a 'response generation' process **14100** where the server generates the response to be sent. The response may be one or more HTML pages, or other World Wide Web resource. Process flow continues to a 'browsing' process **14110** in which the user may browse World Wide Web resources while they complete their call. Process flow continues to the 'outgoing call handling' process **14120** where the IAD/PBX receives the number the user desires to call and establishes the connection. Process flow continues to an 'outgoing call' process **14130** in which the user carries on his or her outgoing call. The 'outgoing call' process **14130** is shown within the 'browsing' process **14110** to indicate that user may operate the browser application during the 'outgoing call' process **14130**. When the 'outgoing call' process **14130** completes process flow completes through an 'end' terminal **14150**.

Fig. 15 depicts a flow diagram for a method for communicating a feedback PopupM message to a computer/phone in conjunction with an incoming communication in accordance with an illustrative embodiment. An 'incoming feedback PopupM' method **15000** initiates at a 'start' terminal **15010** and process flow continues to receive an 'incoming call' input block **15020** whereupon a 'detects incoming call' process **15030** executes. The 'incoming call' input block **15020** is passed to the computer/phone and an 'incoming call indication' process **15040** provides a indication that there is an incoming call, e.g. a conventional ring or simulated ring.

Process flow continues back to the IAD/PBX where a 'send PopupM' decision process **15050** determines whether a PopupM should be sent to the computer/phone.

The 'send PopupM' decision process **15050** may consult a profile in local or remote data store as has been previously illustrated.

If the PopupM should be sent, the 'send PopupM' decision process **15050** exits though its 'yes' branch and process flow continues to a 'PopupM communication' process **15060** where the IAD/PBX retrieves the PopupM and transmits it to the computer/phone where a 'PopupM display' process **15070** presents it to the user.

In contrast to the 'outgoing feedback PopupM' method **14000** when a call is incoming, preferred embodiments of the invention do not provide the user with the option of placing a call to a third party, although this feature could be implemented by one of skill having the benefit of this disclosure. The call to the third party could be placed after completion of the incoming call.

From the 'PopupM display' process **15070** process flow continues to a 'resource identifier selection typing' process **15080** that determines a type of resource selected by the user. If the user does not select a feedback option within a predetermined time, process flow continues to an 'incoming call handling' process **15120** in which the IAD/PBX establishes the connection for the user's incoming call.

If the resource identifier selected by the user is a Uniform Resource Identifier, the 'resource identifier selection typing' process **15080** passes this information to a browser application running on the computer/phone which sends a request message for the identified resource. A 'uniform resource identifier request' input block **15090** is received by the server from which the identified resource is available and process flow continues to a 'response generation' process **15100** where the server generates the response to be sent. The response may be one or more HTML pages, or other World Wide Web resource. Process flow continues to a 'browsing' process **15110** in which the user may browse World Wide Web resources while the incoming call is completed. Process flow completes through an 'end' terminal **15140**.

7. *ALIAS DIALING*

The alias feature will now be descried in more detail. **Fig. 16** depicts a flow diagram an 'alias dialing' method **16000** in accordance with an illustrative embodiment. Process flow initiates at a 'start' terminal **16010** and continues to an 'off hook detection' data block **16020** that contains a signal that a user's aural communication device is "off hook." When the user enters a dialing sequence a 'dialing sequence' input block **16030** is received and is passed to the call management module **3110**. Next a 'recognized alias' decision process **16040** determines whether the dialing sequence corresponds to a recognized alias. In some embodiments, an alias may be recognized by having a "star" before and after the digits, e.g., "*489*", in other embodiments an alias may be recognized by a "star" before the digits coupled with a delay of suitable length, e.g. 5 seconds after entry of the last digit. If an alias is not recognized, the 'recognized alias' decision process **16040** exits through its 'no' branch and process flow completes though an 'end' terminal **16070**. If the call management module **3110** recognizes an alias, the 'recognized alias' decision process **16040** exits through its 'yes' branch and process flow continues to a 'target retrieval' process **16050**. In the 'target retrieval' process **16050**, the call management module **3110** sends the dialing sequence from the 'dialing sequence' input block **16030** to an alias server, which may be, for example, a message server such as the first message server **1110** or the second message server **1120**. The alias server will translate the alias name into a corresponding phone number, generates, and communicates a response message comprising the phone number. When the alias server is also a message server, an informational message may also be returned. The call management module **3110** receives the response message via the message server agent **3030**. Process flow continues to a 'target calling' process **16060** where the call management module **3110** places the call to the phone number received in the response. When an informational message is also received, the 'target calling' process **16060** may also transmit the informational message. In some embodiments the informational message may indicate the party being called, e.g. "Now dialing ITX

Networks," and the user will be connected to the target of the alias. The user may then used their aural communication device conventionally with the target of the alias. Process flow completes through the 'end' terminal **16070**.

All patents, patent applications, documents, standards, protocols, and draft protocols referred to herein are incorporated herein by this reference in their entirety.

Although the present invention has been described in terms of illustrative embodiments, one skilled in the art will understand that various modifications and alterations may be made without departing from the scope of the invention. Accordingly, the scope of the invention is not to be limited to the particular embodiments discussed herein, but should be defined only by the appended claims and equivalents thereof.

Claims

What is claimed is:

1. A method for providing at least one informational message to an aural communication device user in connection with an outgoing communication, said method comprising:

retrieving a first informational message from a message server;

storing said first informational message in a memory;

detecting a first aural communication device is in use;

retrieving said first informational message from said memory;

transmitting said first informational message to said first aural communication device for perception by said user; and

providing an indication to said first aural communication device that said first aural communication device is available for connection to a second aural communication device.

2. A method according to claim 1 wherein said transmitting step initiates before said providing step.

3. A method according to claim 1 wherein said providing step initiates before said transmitting step.

4. A method according to claim 1 wherein said informational message prompts a user of said first aural communication device to respond with said first aural communication device and further comprising:

receiving a response from the user with said first aural communication device.

5. A method according to claim 4 wherein said informational message comprises a location.

6. A method according to claim 5 further comprising:

establishing a communication path between said first aural communication device and said location upon receiving said response from the user of said first aural communication device.

7. A method according to claim 6 wherein at least a portion of said communication path is established over a local area network..

8. A method according to claim 6 wherein said location is a phone number and said communication path is established to a second aural communication device.

9. A method according to claim 5 wherein said location is uniform resource identifier.

10. A method according to claim 4 wherein said response is an alias.

11. A method according to claim 10 further comprising:

querying a database to map said alias to a location; and

establishing a communication path between said first aural communication device and said location upon receiving said response from the user said first aural communication device.

12. A method according to claim 1 wherein said first aural communication device comprises a telephone.

13. A method according to claim 1 wherein said first aural communication device comprises an etherphone.

14. A method according to claim 1 wherein said first aural communication device comprises a computer, a microphone, a speaker, and a computer telephony application.

15. A method according to claim 1 wherein computer telephony application comprises a packetized voice communications application.

16 A method according to claim 1 wherein said packetized voice communications application is a VoIP application.

17. A method according to claim 1 wherein said first aural communication device comprises wireless communication unit.

18. A method according to claim 1 wherein said wireless communication unit comprises a cellular communications unit.

19. A method according to claim 1 wherein said informational message provides the user of said aural communication device with a selectable option.

20. A method according to claim 19 further comprising:

receiving an indication of the selection of said selectable option, and

communicating said indication of the selection of said selectable feedback option to message sever, wherein thereafter an automated response is generated.

21. A method according to claim 20 wherein said automated response is an electronic mail message.

22. A method according to claim 20 wherein said automated response is a coupon.

23. A method according to claim 20 wherein said automated response is communicated to a merchant.

24. A method according to claim 1 wherein said informational message is selected responsive to a profile associated with said user.

25. A method according to claim 1 wherein said informational message is selected responsive to a profile associated with a sponsor.

26. A method according to claim 1 wherein said step of detecting a first aural communication device is in use is performed by a first switching system, and further comprising:

receiving routing information;

routing communications from said first aural communication device to a second switching system responsive to said routing information; and wherein the additional steps of claim 1 are performed by said second switching system.

27. A method according to claim 26 wherein said routing information comprises:

a routing code.

28. A method according to claim 27 wherein said routing code comprises:

a primary interexchange carrier (PIC) code.

29. A method according to claim 27 wherein said routing code comprises:

a predetermined number of dialing digits detected by a service control point as a subscriber-based routing trigger.

30. A method according to claim 26 wherein said routing information comprises:

an off-hook condition detected by a service control point as a subscriber-based routing trigger.

31. A method according to claim 3 further comprising:

determining if said second aural communication device has answered; and

if said second aural communication device has answered, indicating the establishment of the connection; and

transmitting a second informational message to said first aural communication device for perception by said user.

32. A method according to claim 31 wherein said second informational message is acoustically mixed with a ring back signal.

33. A method according to claim 3 wherein if said second aural communication device has not answered, performing at least one time steps of: retrieving a second informational message and transmitting said second informational message to said first aural communication device for perception by said user

34. A method according to claim 31 wherein said second informational message is selected responsive to a profile of a sponsor.

35. A method according to claim 1 wherein said first informational message is compressed and the step of transmitting said first informational message to said first aural communication device for perception by said user comprises:

decompressing said first informational message; and

transmitting said decompressed first informational message.

36. A method according to claim 1 wherein said first informational message is in text form and the step of transmitting said first informational message to said first aural communication device for perception by said user comprises:

converting said text form to an audio form; and

transmitting said audio form of said first informational message.

37. A method according to clai 1 further comprising:

authenticating the identity of said user.

38. A method according to claim 37 wherein authenticating the identity of said user comprises receiving personal identification information from said user.

39. A method according to claim 38 wherein said personal identification information comprises a voice print.

40. A method for providing at least one informational message to an aural communication device user in connection with an incoming communication, said method comprising:

receiving an indication of an incoming communication from a first aural communication device to a second aural communication device;

detecting said second aural communication device has answered said incoming communication;

retrieving a first informational message; and

transmitting said first informational message to said second aural communication device for perception by the user of said second aural communication device, wherein said transmitting step follows said detecting step.

41. A method according to claim 40 wherein said informational message is selected responsive to a profile associated with said user.

42. A method according to claim 40 wherein said informational message is selected responsive to a profile associated with a sponsor.

43. A method according to claim 40 further comprising:

providing a ringback signal to said first aural communication device;

establishing a communication path to said first aural communication device; and performing steps comprising:

retrieving a second informational message; and

transmitting said second informational message to said first aural communication device for perception by the user of said first aural communication device,

at least one time.

44. A method according to claim 40 wherein said first informational message is compressed and the step of retrieving a first informational message further comprises:

decompressing said first informational message; and

transmitting said decompressed first informational message.

45. A method according to claim 40 wherein said first informational message is in text form and the step of retrieving a first informational message further comprises converting said text form to an audio form and transmitting said audio form of said first informational message.

46. An apparatus configured for connection to at least one aural communication device and a data network, said apparatus for providing at least one informational message to a user of an aural communication device, said apparatus comprising:

a processor;

a memory communicatively coupled with said processor;

a network interface communicatively coupled with said processor, said network interface configured for communication over said data network;

a subscriber line interface communicatively coupled with said processor; and

a message agent, said message agent adapted to retrieve at least one informational message for storage in said memory, wherein said at least one message may be communicated to said subscriber line interface under the control of said processor.

47. An apparatus according to claim 46 wherein said subscriber line interface communicatively coupled with said processor comprises:

a line interface circuit, said line interface circuit configured to detect if said aural communication device connected to said apparatus is in use, said line interface circuit further configured to provide an incoming communication signal to said aural communication device, said line interface circuit further configured to transmit and receive telecommunications signals to said aural communication device; and

a line audio processor, said line audio processor configured to convert audio signals to digital samples for communication to said line interface circuit.

48. A method for transmitting at least one informational message to a packetized voice client for perception by a user of said packetized voice client comprising:

receiving an indication of an outgoing packetized voice communication from a sender client to a recipient client;

sending an indication to said sender that said outgoing packetized voice communication was accepted by said recipient client;

transmitting an informational message to said sender for perception by a user of said sender client; and

thereafter forwarding an indication of an outgoing packetized voice communication to said recipient client, wherein said recipient client may accept said outgoing communication.

49 A method according to claim 48 wherein said informational message prompts said user of said sender client to respond with said sender client and future comprising:

receiving a response from the user with said sender client.

50. A method according to claim 49 wherein said informational message provides the user of said aural communication device with a selectable feedback option.

51. A method according to claim 50 further comprising:

receiving an indication of the selection of said selectable feedback option, and

communicating said indication of the selection of said selectable feedback option to a message sever, wherein thereafter an automated response is generated.

52. A method for transmitting at least one informational message to a packetized voice client for perception by a user of said packetized voice client comprising:

receiving an indication of an acceptance of a packetized voice communication from a recipient client, said acceptance communicated responsive to a request to initiate a packetized voice communication from a sender client;

transmitting an informational message to said recipient client for perception by a user of said recipient client; and thereafter;

forwarding said acceptance to said sender client.

53. A method according to claim 52 wherein said informational message is selected responsive to a profile associated with said user.

54. A method according to claim 52 wherein said informational message is selected responsive to a profile associated with a sponsor.

55. A system for providing interactive informational messages to user of aural communication device comprising:

a telephone switching system, said telephone switching system further comprising a message agent, a network interface, and a subscriber line interface, said network interface configured for communicative coupling with a telecommunications network; and

a networked computer system, said computer comprising a programmable computer, a display, a microphone, a speaker, and a telecommunications interface, said telecommunications interface communicatively coupled with said subscriber line interface of said telephone switching system, said computer system configurable to provide an aural communication device by receiving aural communications with said microphone and audibly reproducing aural communications with said speaker,

instructions configuring said programmable computer to provide a client application, instructions further configuring said computer to display an informational message responsive to signals from said telephone switching system.

56. A system according to claim 55 wherein said informational message comprises a selectable action.

57. A system according to claim 56 wherein selection of said selectable action comprises sending a request for a resource available on the data network.

58. A system according to claim 56 wherein selection of said selectable action comprises initiating a telephonic communication.

59. A method for providing an informational message in conjunction with a computer-telephony system, said computer-telephony system comprising a computer, a speaker, and a microphone, said speaker and microphone operatively coupled with said computer, programmed instructions configuring said computer-telephony system to provide a first aural communication device, said method comprising:

transmitting at least one informational message to said computer telephony system;

detecting said first aural communication device is in use; and

providing a signal to said computer system to present one of said at least one informational message, wherein thereafter said one of said at least one informational message is displayed for perception by a user of said first aural communication device.

60. A method according to claim 59 wherein said informational message is selected responsive to a profile associated with said user.

61. A method according to claim 59 wherein said informational message is selected responsive to a profile associated with a sponsor.

62. A method according to claim 59 wherein said informational message comprises a selectable resource identifier, said selectable resource identifier associated with a communication resource.

63. A method according to claim 62 further comprising:

receiving an indication of the selection of said resource identifier;

establishing a communication path between said first aural communication device and said communication resource associated with said selected resource identifier.

64. A method according to claim 63 wherein said selectable resource identifier is a uniform resource identifier and said communication resource is a world wide web server.

65. A method according to claim 63 wherein said selectable resource identifier is phone number and said communication resource is a second aural communication device.

66. A method according to claim 59 wherein said informational message comprises a first selectable resource identifier associated with a first communication resource and a second selectable resource identifier associated with a second communication resource, said method further comprising:

receiving an indication of the selection of said first resource identifier;

establishing a communication path between said first aural communication device and said first communication resource associated with said first selected resource identifier.

receiving an indication of the selection of said second resource identifier;

establishing a communication path between said first aural communication device and said second communication resource associated with said second selected resource identifier.

67. A system for providing informational messages to a user of aural communication device comprising:

a switching system, said switching system routing voice telecommunications between at least a first aural communication device and a second aural communication device, said switching system further configured for communicative coupling with a data network;

a message server, said message server configured for client-server communications with said switching system via said data network, said message server storing an informational message;

a message manager, said message manager configuring said switching system to request said informational message from said message server; and

a call manager, said call manager configuring said switching system to perform steps comprising:

receiving said informational message from said message manager; and

transmitting said informational message to said first aural communication device.

68. A system according to claim 67 wherein said switching system is further configured receive calls routed from a central switch, said central switch configured to route calls to said system responsive to routing information comprising:

a routing code.

69. A system according to claim 68 wherein said routing code comprises:

a a primary interexchange carrier (PIC) code.

70. A system according to claim 68 wherein said routing code comprises:

a predetermined number of dialing digits detected by a service control point as a subscriber-based routing trigger.

71. A system according to claim 67 wherein said routing information comprises:

an off-hook condition detected by a service control point as a subscriber-based routing trigger.

72. A system according to claim 67 wherein said call manager further configures said switching system to perform steps comprising:

receiving dialing information from said first aural communication device, said dialing information for establishing a communications path to said second aural communication device;

receiving an off hook indication from said second aural communication device;

transmitting a second informational message to said second aural communication device; and

establishing a communication path between said first aural communication device and said second aural communication device.

73. A system according to claim 67 further comprising an interactive voice response ("IVR") system for speaker verification, said IVR system configured for communication with said central switching system.

74. A system according to claim 67 wherein said call manager performs steps further comprising:

detecting said first aural communication device is in use; and

providing a dial tone to said first aural communication device subsequent to transmitting said informational message.

75. A system according to claim 67 wherein said call manager performs steps further comprising:

detecting said first aural communication device is in use; and

providing a dial tone to said first aural communication device prior to transmitting said informational message.

76. A system according to claim 67 wherein said call manager performs steps further comprising:.

receiving an off hook signal from said first aural communication device;

receiving dialing information from said first aural communication device, said dialing information for establishing a communications path to said second aural communication device;

receiving an off hook signal from said second aural communication device; and

providing a dial tone to said first aural communication device prior to transmitting said informational message.

77. In a wireless communications network, a method for providing an informational message in conjunction with an incoming communication for an aural

communication device user, said aural communication device having an associated identifier, said method comprising:

receiving an indication of said incoming communication for said aural communication device;

retrieving an informational message from a message server;

establishing a data communication connection to said aural communication device;

transmitting said informational message to said aural communication device via said data communication connection; and

thereafter establishing a voice connection to said aural communication device for said incoming communication.

78. A method according to claim 77 wherein said informational message comprises image data.

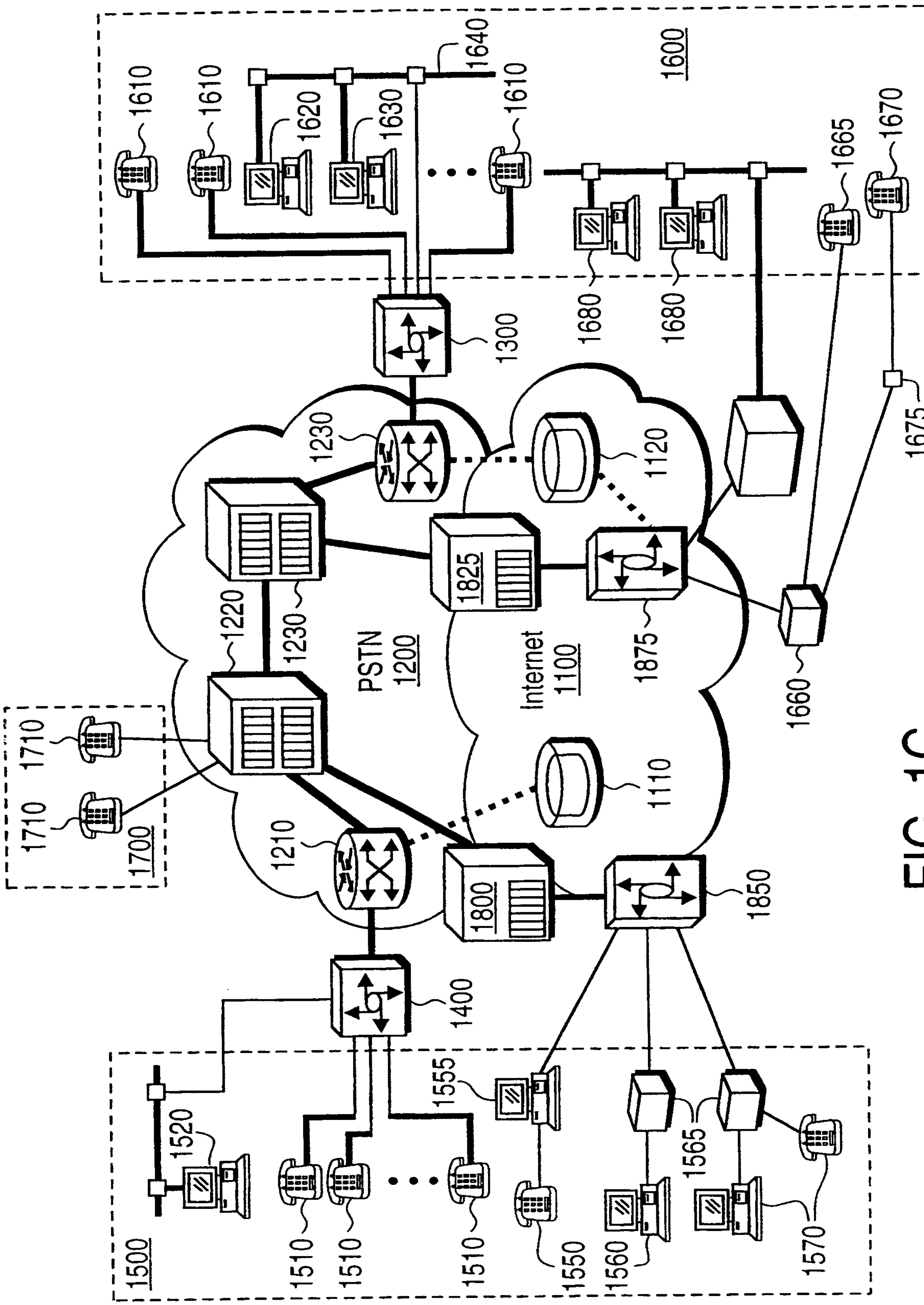
79. A method according to claim 77 wherein said informational message comprises textual data.

80. A method according to claim 77 wherein authenticating the identity of said user comprises receiving personal identification information from said user.

81. A method according to claim 80 wherein said personal identification information comprises a voice print.

82. A method according to claim 77 wherein retrieving an informational message from a message server comprises:

providing said identifier of said aural communication device to a message server;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1510
1510
1510
1550
1560
1570
1500

FIG. 1C

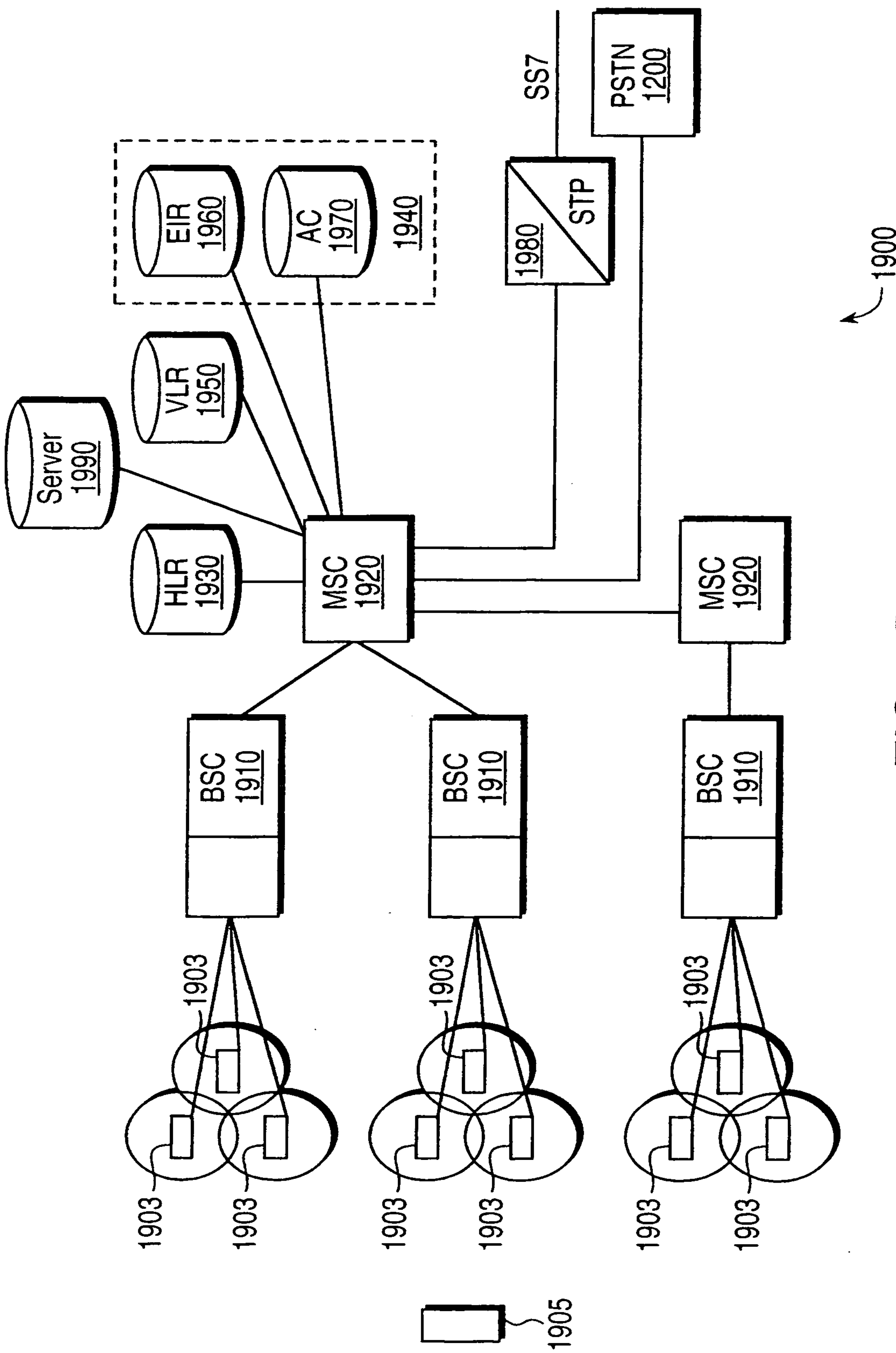

SS7
PSTN
1200
EIR
1960
AC
1970
1940
1980
STP
1900
VLR
1950
Server
1990
HLR
1930
MSC
1920
MSC
1920
BSC
1910
BSC
1910
BSC
1910
1903
1903
1903
1903
1903
1903
1903
1903
1903
1905

FIG. 1D

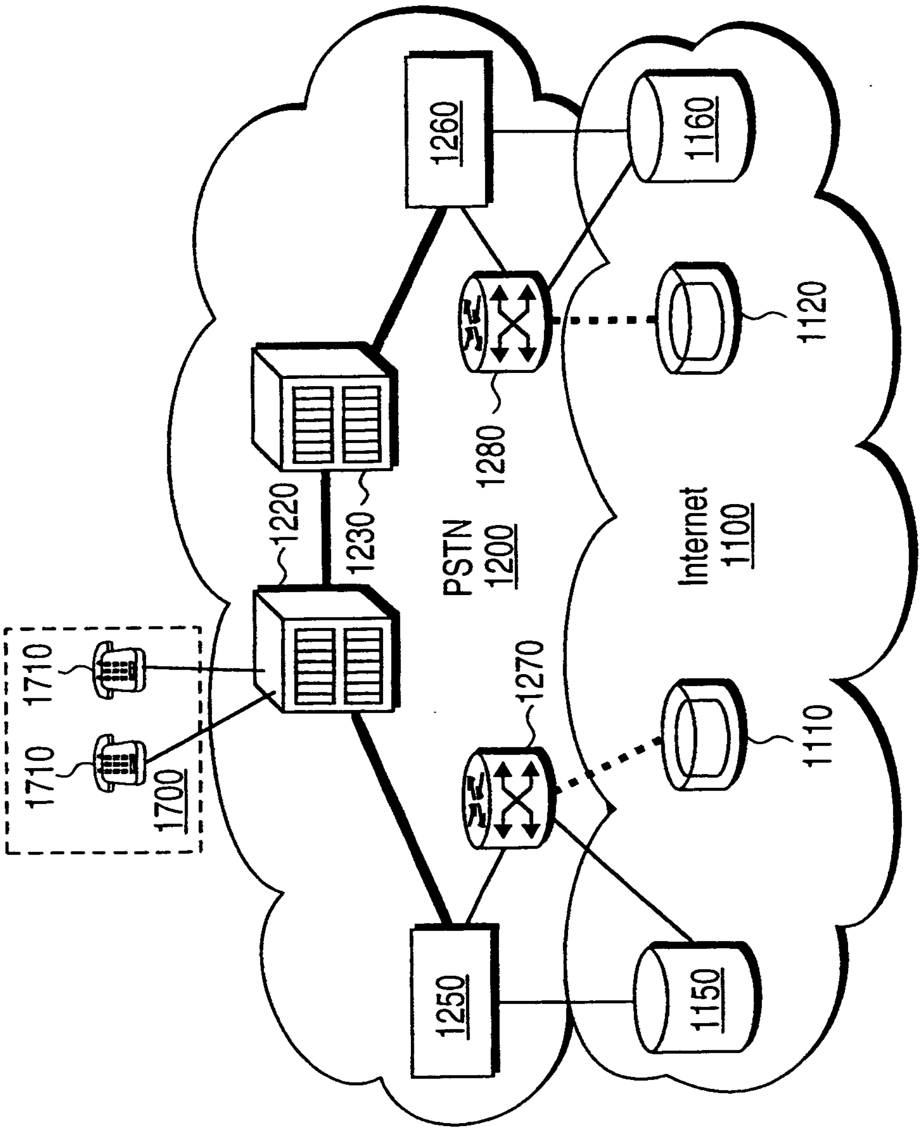

1260
1160
1120
1280
1220
1230
PSTN
1200
Internet
1100
1710
1710
1700
1270
1110
1250
1150

FIG. 1E

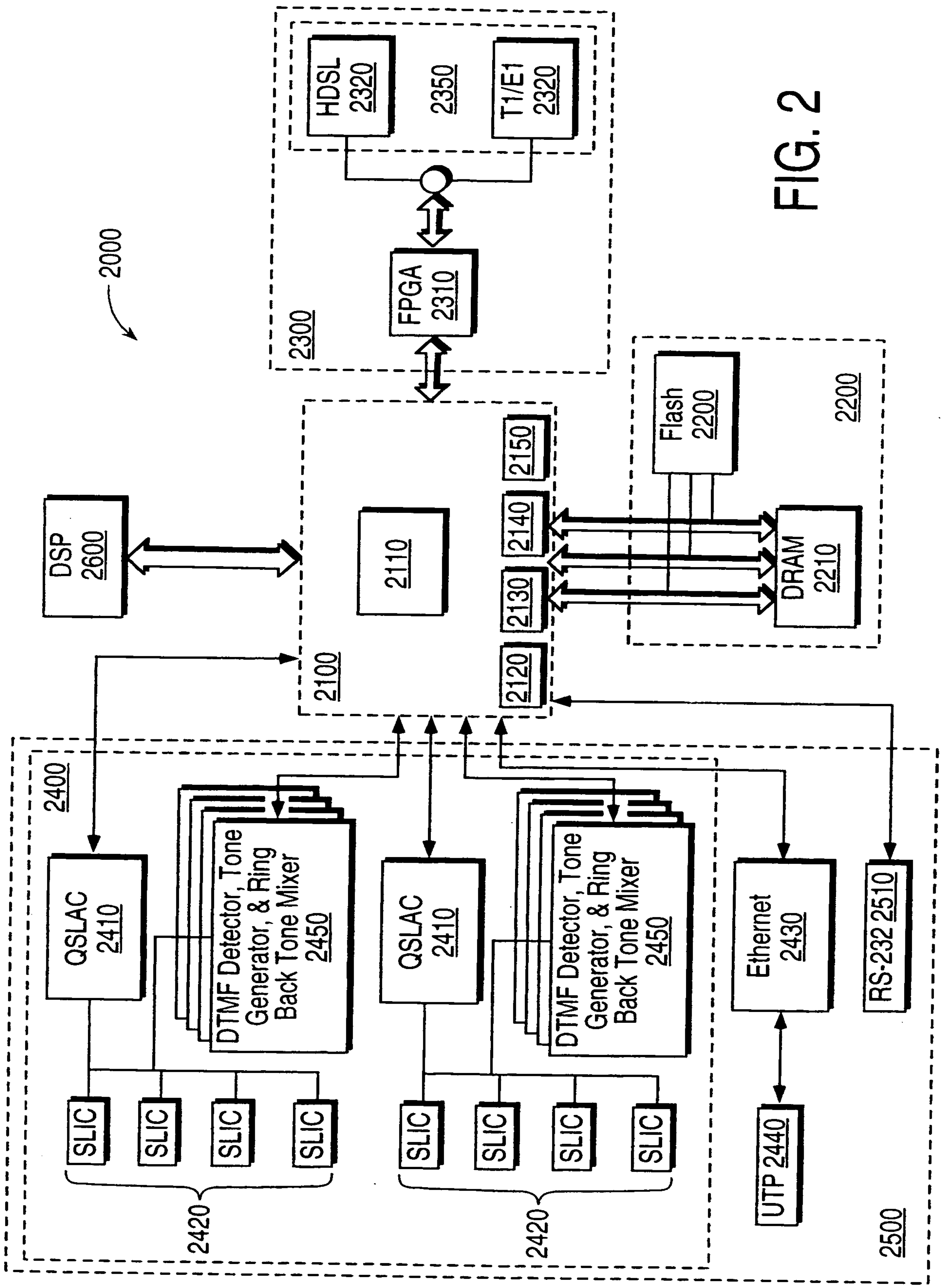
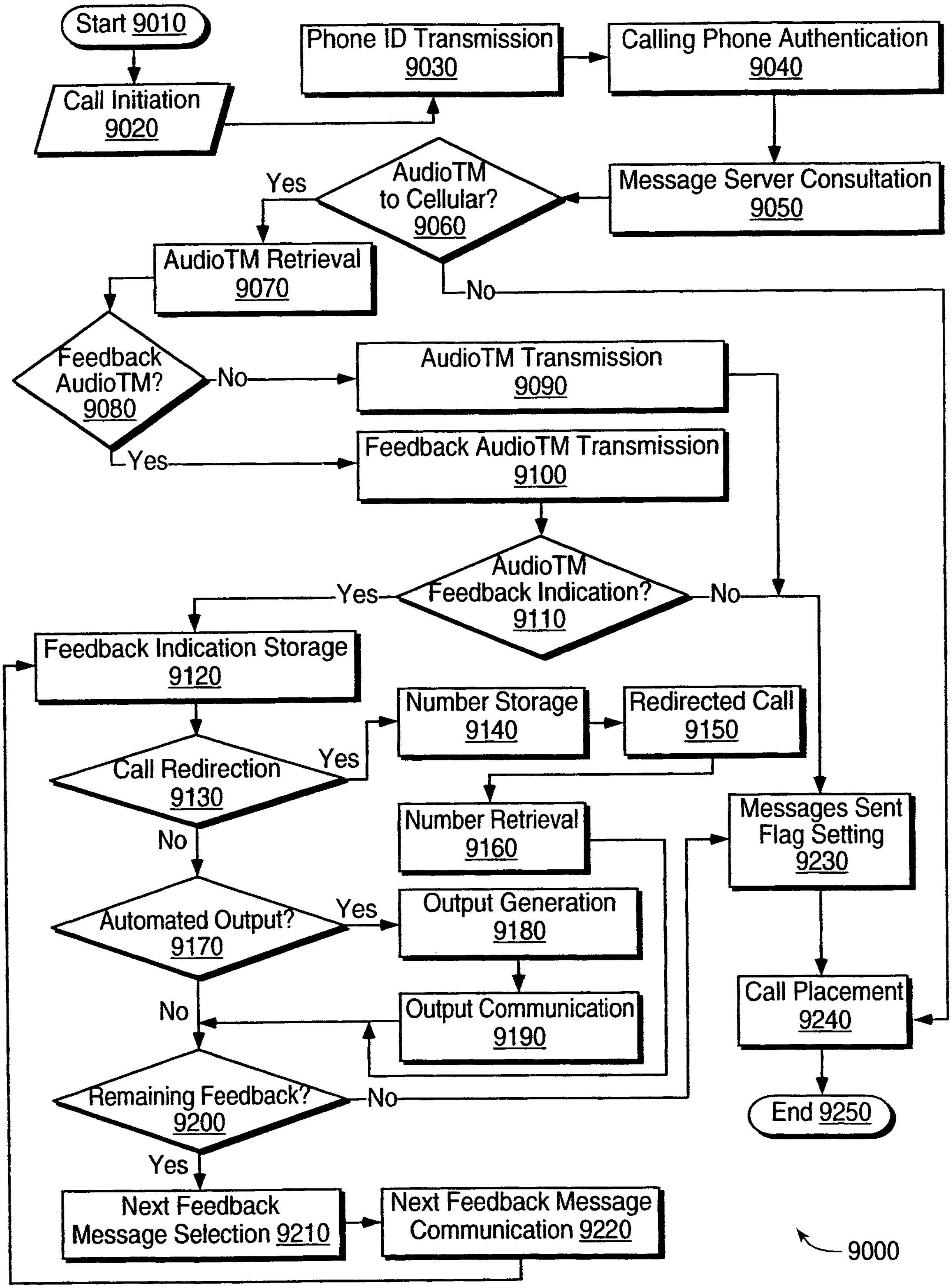

2000
2300
HDSL
2320
2350
T1/E1
2320
FPGA
2310
2200
Flash
2200
DRAM
2210
DSP
2600
2110
2100
2150
2140
2130
2120
2400
QSLAC
2410
DTMF Detector, Tone Generator, & Ring Back Tone Mixer
2450
QSLAC
2410
DTMF Detector, Tone Generator, & Ring Back Tone Mixer
2450
Ethernet
2430
RS-232 2510
SLIC
SLIC
SLIC
SLIC
SLIC
SLIC
SLIC
SLIC
2420
2420
UTP 2440
2500

FIG. 2

FIG. 9


Start 9010
Call Initiation 9020
Phone ID Transmission 9030
Calling Phone Authentication 9040
Message Server Consultation 9050
AudioTM to Cellular? 9060
Yes
No
AudioTM Retrieval 9070
Feedback AudioTM? 9080
No
Yes
AudioTM Transmission 9090
Feedback AudioTM Transmission 9100
AudioTM Feedback Indication? 9110
Yes
No
Feedback Indication Storage 9120
Call Redirection 9130
Yes
No
Number Storage 9140
Redirected Call 9150
Number Retrieval 9160
Automated Output? 9170
Yes
No
Output Generation 9180
Output Communication 9190
Remaining Feedback? 9200
No
Yes
Next Feedback Message Selection 9210
Next Feedback Message Communication 9220
Messages Sent Flag Setting 9230
Call Placement 9240
End 9250
9000

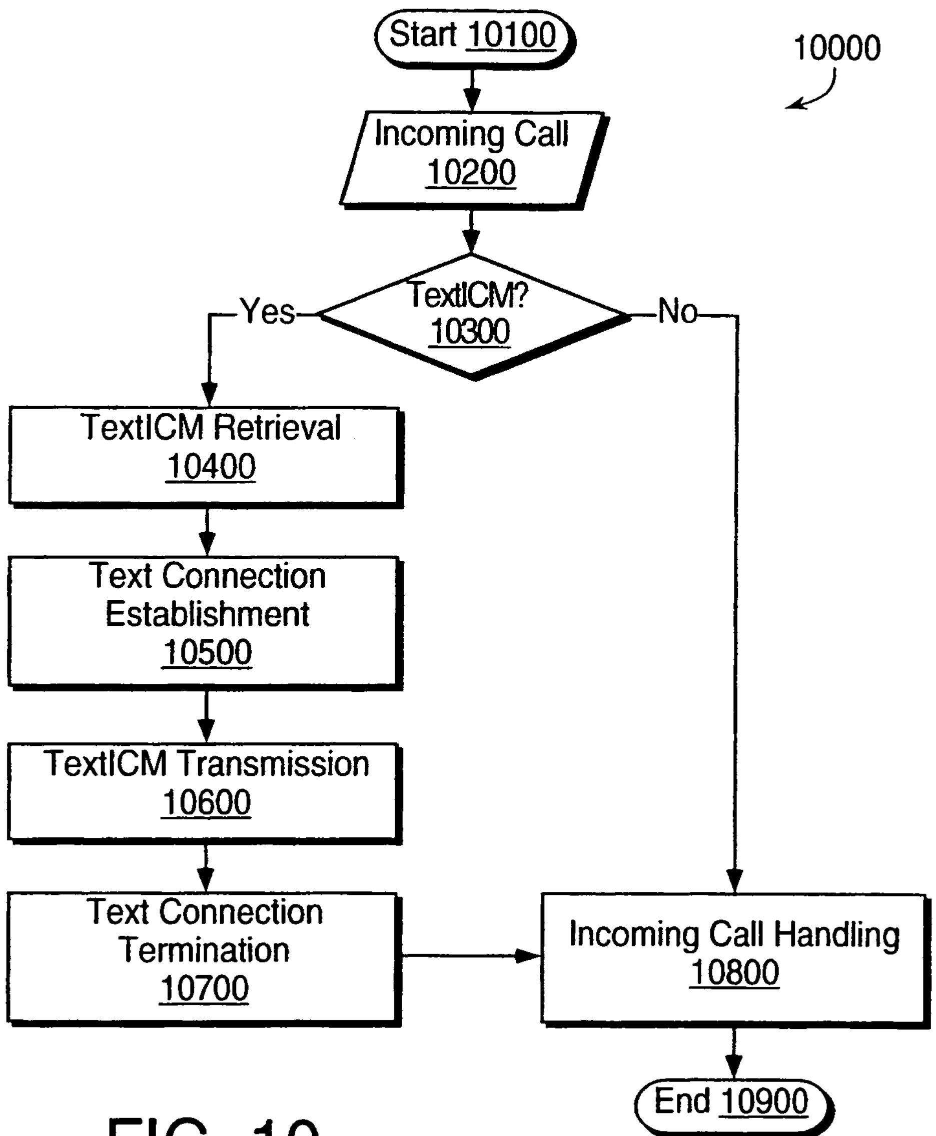

Start 10100
10000
Incoming Call 10200
TextICM? 10300
Yes
No
TextICM Retrieval 10400
Text Connection Establishment 10500
TextICM Transmission 10600
Text Connection Termination 10700
Incoming Call Handling 10800
End 10900

FIG. 10

FIG. 11

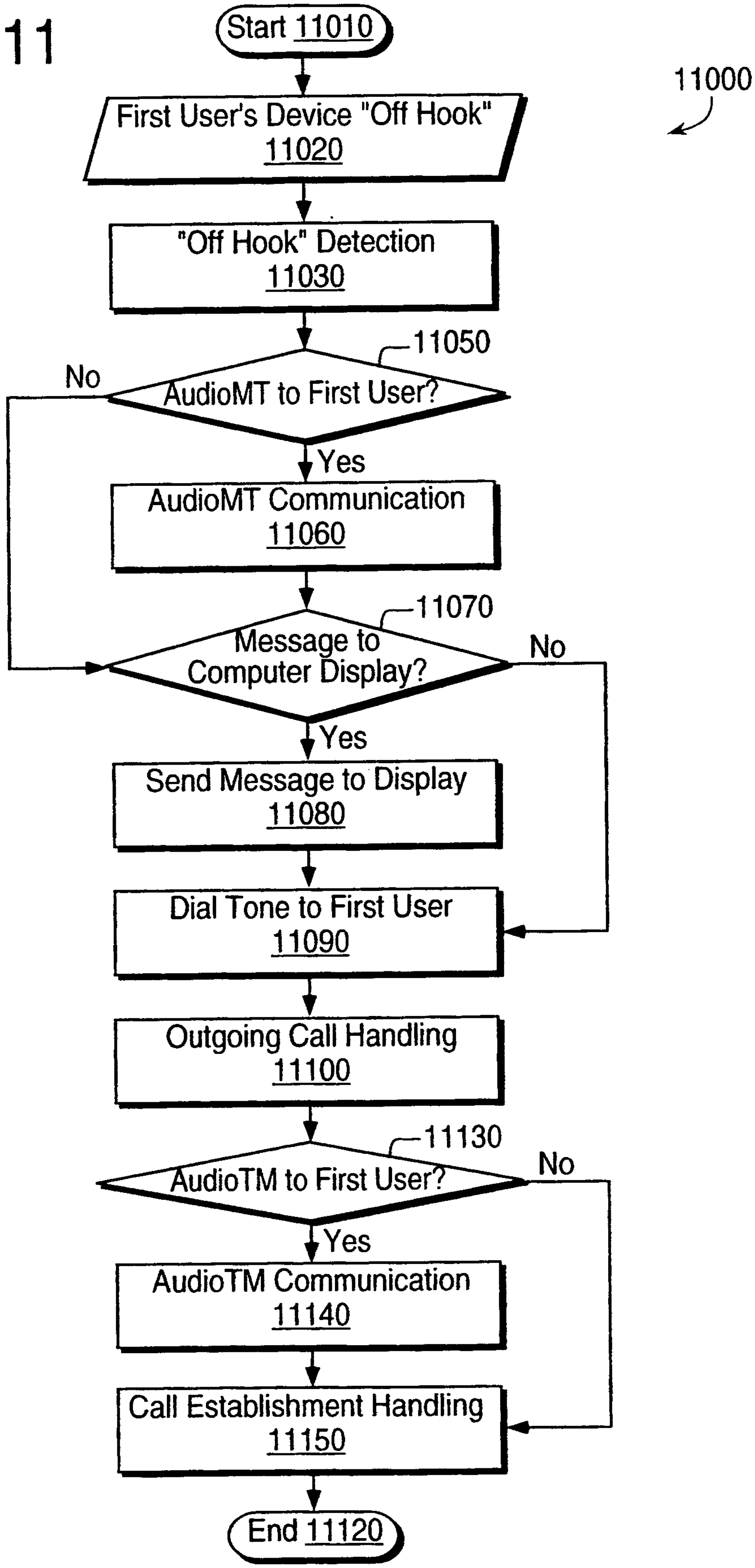

11000
Start 11010
First User's Device "Off Hook" 11020
"Off Hook" Detection 11030
11050
AudioMT to First User?
No
Yes
AudioMT Communication 11060
11070
Message to Computer Display?
No
Yes
Send Message to Display 11080
Dial Tone to First User 11090
Outgoing Call Handling 11100
11130
AudioTM to First User?
No
Yes
AudioTM Communication 11140
Call Establishment Handling 11150
End 11120

FIG. 12

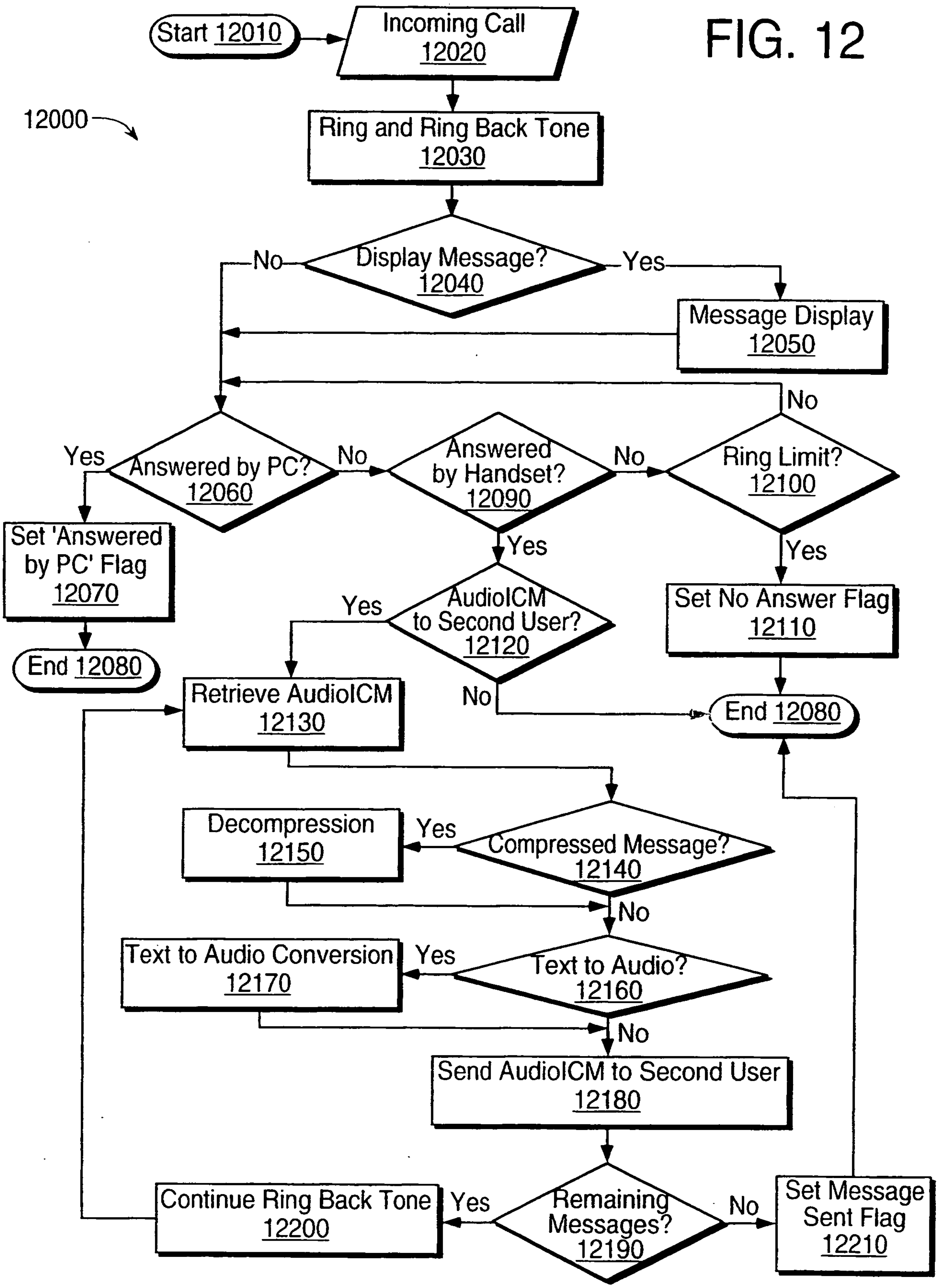

12000
Start 12010
Incoming Call 12020
Ring and Ring Back Tone 12030
Display Message? 12040
No
Yes
Message Display 12050
No
Answered by PC? 12060
Yes
No
Answered by Handset? 12090
No
Ring Limit? 12100
Yes
Set 'Answered by PC' Flag 12070
End 12080
Yes
AudioICM to Second User? 12120
Yes
Yes
Set No Answer Flag 12110
Retrieve AudioICM 12130
No
End 12080
Decompression 12150
Yes
Compressed Message? 12140
No
Text to Audio Conversion 12170
Yes
Text to Audio? 12160
No
Send AudioICM to Second User 12180
Continue Ring Back Tone 12200
Yes
Remaining Messages? 12190
No
Set Message Sent Flag 12210

FIG. 13A

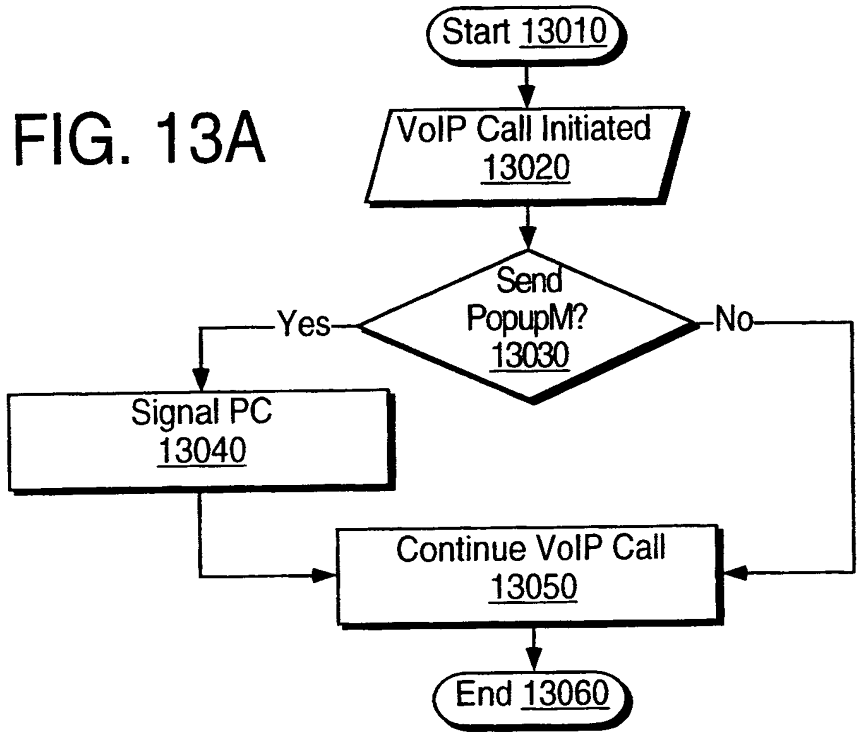
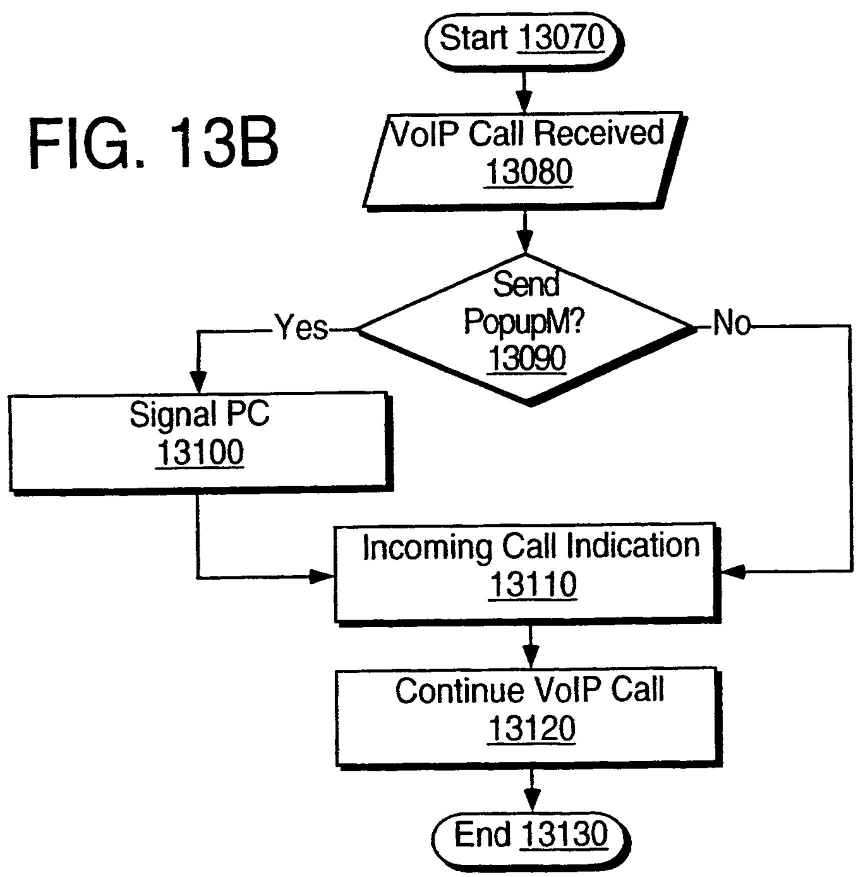


FIG. 13B



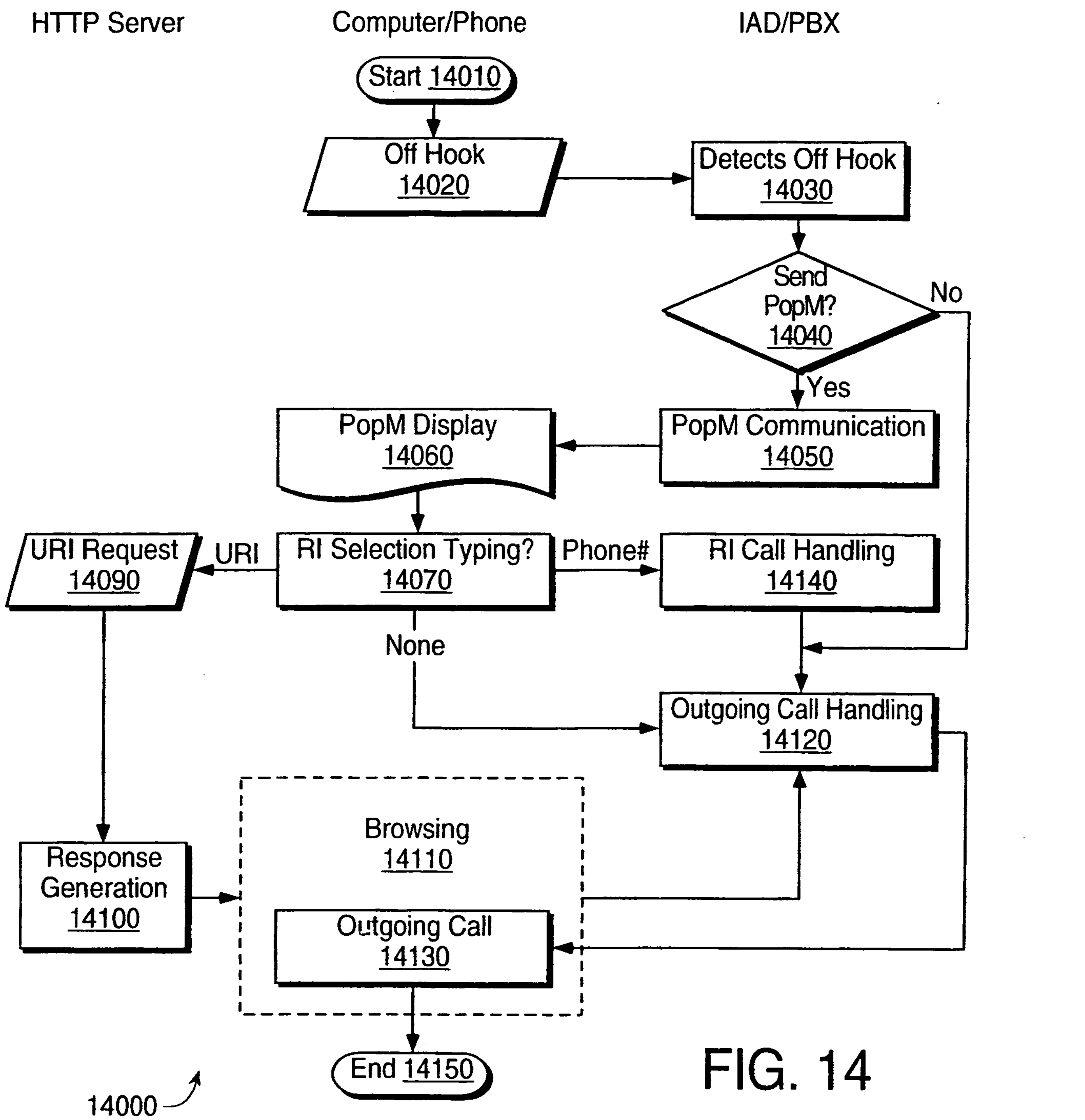

HTTP Server
Computer/Phone
IAD/PBX
Start 14010
Off Hook 14020
Detects Off Hook 14030
Send PopM? 14040
No
Yes
PopM Communication 14050
PopM Display 14060
RI Selection Typing? 14070
URI
URI Request 14090
Phone#
RI Call Handling 14140
None
Outgoing Call Handling 14120
Browsing 14110
Outgoing Call 14130
Response Generation 14100
End 14150
14000

FIG. 14

INTERNATIONAL SEARCH REPORT

International Application No
PCT/US 00/08715

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7 H04M3/487 H04M7/00 H04Q7/22

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 7 H04M H04Q

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

EPO-Internal, WPI Data, INSPEC, PAJ

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	GB 2 206 265 A (AMERICAN TELEPHONE & TELEGRAPH) 29 December 1988 (1988-12-29)	1-6,8,9,12,19,20,23-27,29,40
A	page 2, line 22 - line 32 page 5, line 4 - line 14 page 7, line 7 - line 10 ---	46,48,52,55,59,67,77,84,93
X	US 5 652 784 A (STEINVORTH RANDOLPH H ET AL) 29 July 1997 (1997-07-29) column 1, line 36 - line 46 abstract ---	1-3

-/--

[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

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"A" document defining the general state of the art which is not considered to be of particular relevance

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"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

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"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

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"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
19 September 2000	27/09/2000

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	Vandevenne, M

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INTERNATIONAL SEARCH REPORT

International Application No
PCT/US 00/08715

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication,where appropriate, of the relevant passages	Relevant to claim No.
X	WO 91 06187 A (DUFOUR JEAN PIERRE) 2 May 1991 (1991-05-02) page 5, line 31 - line 37	40
X	US 5 448 625 A (LEDERMAN MATTHEW) 5 September 1995 (1995-09-05) column 5, line 46 -column 7, line 67	40,42,43
X	WO 98 32275 A (PALMAEUS FREDERIK ;ANDER CARL (SE); GRATISTELEFON SVENSKA AB (SE)) 23 July 1998 (1998-07-23) page 5, line 29 -page 7, line 10	40-43
A	WO 98 36585 A (NORTHERN TELECOM INC) 20 August 1998 (1998-08-20) abstract	
A	WO 98 32271 A (MARTUCCI GIUSEPPE ;BRITISH TELECOMM (GB)) 23 July 1998 (1998-07-23) abstract	
A	LOW C: "THE INTERNET TELEPHONY RED HERRING" HP LABORATORIES TECHNICAL REPORT,XX,XX, no. 96/98, 15 May 1996 (1996-05-15), pages 1-15, XP002043669 paragraph ‘0006!	
A	US 5 850 433 A (RONDEAU DAVID EDWARD) 15 December 1998 (1998-12-15) abstract	

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INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No
PCT/US 00/08715

Patent document cited in search report		Publication date	Patent family member(s)			Publication date
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			AU	607233	B	28-02-1991
			AU	1838588	A	05-01-1989
			CA	1286759	A	23-07-1991
			JP	1017553	A	20-01-1989
			JP	1764247	C	28-05-1993
			JP	4058221	B	16-09-1992
US 5652784	A	29-07-1997	NONE			
WO 9106187	A	02-05-1991	FR	2653623	A	26-04-1991
			EP	0450045	A	09-10-1991
US 5448625	A	05-09-1995	CA	2121102	A	14-10-1994
WO 9832275	A	23-07-1998	AU	5684798	A	07-08-1998
WO 9836585	A	20-08-1998	EP	0962090	A	08-12-1999
WO 9832271	A	23-07-1998	AU	5490198	A	07-08-1998
			EP	0953250	A	03-11-1999
US 5850433	A	15-12-1998	NONE			